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CONTRIBUTION OF PROF. I. MALECKI TO THE POLISH SCHOOL OF ACOUSTICS

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A detailed study of the history of the Polish acoustics does not exist. The easiest way to present such a history might consist in describing the activity of separate teams, gathered around several leading personalities. The following publication is an attempt to present the scientific contribution to Polish acoustics of Professor Ignacy Malecki himself and of the teams which were co-operating with him.

The first Polish research on acoustics was started in the middle of the thirties. During the war period, lectures on acoustics were given at the underground Warsaw Technical University. When the country was being rebuilt after the war, solutions of the problems of acoustics of the reconstructed halls, as well as proper solutions for the noise control in the building industry were required. The scientific research at the Warsaw Technical University, and at the Building Research Institute has started. The Electro-acoustical Chair was founded at the Warsaw Technical Unversity. Later on the Institute of Fundamental Technological Research of the Polish Academy of Science was established, where research on acoustics, aimed mainly at the ultrasonic testing, the improvement of noise measurements, and the acoustics of speech, were conducted. Theoretical studies on the electro-mechano-acoustical analogies and the coupled electro-acoustical fields were carried out. Heterogeneous media, especially ceramics, were tested theoretically and experimentally. An interest in hypersounds and in the quantum representation of the acoustical phenomena was growing. The diffraction field and the transport of the vibrations through the ear channel have been described using the quantum method. Focal point hase been shiflet from the measurement of the acoustic waves velocity and their attenuation to the investigation of the active phenomenon, i.e. the acoustic emission was performed, and different fields of its application were developed. The acoustic emission during the brittle fracture and during chemical reactions were also studied. Scientific cooperation with other countries and the teaching activity associated with the research undertaken were conducted.

1. The pre-war period

The first research and technical work on acoustics in Poland date from 1935–36. At that time, a few centres, which dealt with those problems, were set up. These were: the Institute of Physics of the Poznań University, (M. Kwiek), the Research

Department of the State Tele- and Radioengineering, Warsaw (Z. Żyszkowski), the State Institute of Telecommunication (T. Korn, W. Pajewski). The institution which was mostly interested in the development of acoustics in Poland was, however, the "Polish Radio", therefore the foundation of the Technical Division of the Polish Radio Research Department in 1936 was the fact of a great importance. Engineer Ignacy Małecki (born 18th of November 1912) was appointed the head of this division; in that time he had already some experience in the field of acoustics, since in 1935 he had started his doctor's thesis, promoted by a famous acoustician Prof. Edwin Mayer, at the Heinrich Hertz Institut in Berlin. The thesis was dealing with generation of ultrasounds of very high frequency, as it was regarded in these times (200 kHz), by a tourmaline crystal. The political course of events interrupted his work at the Institute, so that I. Malecki came back home and took up the above-mentioned post.

At those times, the Polish Radio used almost exclusively foreign electro-acoustical equipment, mainly German and American. Design of the broadcasting equipment in Poland was therefore not needed, but the improvement of the sound quality in broadcasting studios by their suitable shaping, and proper selection and arrangement of microphones were of main interest. I. Malecki was acquainted with the broadcasting technique in certain European broadcasting stations, so he could undertake the task in a competent way, which consisted in designing the acoustic system for the Polish Radio Center. The Center was to be founded in the very down-town area of Warsaw. The design of the Center was completed before the war, and the work on foundations of its main building started in the spring in 1939. The design contained many pioneering (in those times) solutions concerning, among others, the technological process of recording, and the wide range regulation of the reverberation time of the studios.

2. The Second World War (1939–1944)

The outbreak of the Second World War cancelled these plans. Vicissitudes of Polish acousticians during the war were different. Some of them left the country, others, as M. Kwiek, earned his living by repairing church organs. I. Malecki stayed in Warsaw. As he cooperated with the Warsaw Technical University before the war, he took part in organizing of the underground activity of this college. For him, it was a possibility to do some teaching, though in a limited range, on acoustics. He lectured on the architectonic acoustics at the Achitecture Faculty of the underground Technical University. Using professional literature smuggled from abroad, he continued theoretical studies related with his previous experimental research, which had been undertaken during his work for the Polish Radio. Working often with the risk of his life, he completed his doctor's thesis, which he defended at the underground University in 1941, and in 1943 he presented his habilitation thesis (needed for the position of an associate professor), and he received the "veniam legendi" (the right to lecture at the universities). Both theses referred to the architectural acoustics; they were entitled: '*The Physics of Porous Materials*' [1], in which he dealt with the mechanism of attenuation of sonic waves by capillary structures, and *Propagation of Acoustic Waves* *in Halls*. In the latter thesis he developed the theory of acoustic field in case of an uneven distribution of sound absorbing materials on the walls of a room, and he applied a statistical approach to transient states in a bounded space. I. Malecki came there independently to the same conclusions which may be found in the results of the research of L. Cremer and R.H. Bolt, which were undertaken at the same time.

3. The 1945–1953 period

Remnants of the laboratories of the Warsaw Technical University were destroyed during the Warsaw Uprising in 1944. Some of the few scientists, who survived the war and did not leave the country, started to rebuild the universities and research laboratories. The acoustical institution at the Technical University of Poznań (M. Kwiek) and the laboratories of the State Institute of Telecommunication (J. Kacprowski, W. Pajewski) associated with acoustics were restored. The laboratories were not totally destroyed, since the battles of the Warsaw Uprising did not reach the Institute area. Departments of electro-acoustics (Z. Żyszkowski) and otolaryngology were created in Wrocław from the very foundations.

Since no experimental basis for a scientific work existed then in Warsaw, and the work on reactivation of the Warsaw Technical University proceeded slowly, Dr. I. Malecki decided to leave for Gdańsk. The activity of Prof. Malecki in Gdańsk was characterized in the paper of M. Sankiewicz and G. Budzyński [2]. Prof. Malecki headed a group of people who had never dealt with acoustics before. There were: J. Góra, the first postgraduate student of Prof. Malecki, and J. Wojciechowicz, an excellent designer. This group undertook ambitious tasks, which were urgently needed for rebuilding the destroyed country; acoustic designs of the reconstructed and new auditoria, and reproduction of the hydroacoustic equipment used by the German Kriegsmarine.

At the same time education of young engineering staff began. In 1946, at the age of thirty-four, Dr. I. Malecki was nominated a full professor by the president of Poland. In 1950 Prof. Malecki returned to his mother college — the Warsaw Technical University, where he took over the Chair of Electro-acoustics of the Telecommunication Faculty. During a few years he held responsible posts at the University — as the dean of the faculty, and then as the vice-chancellor for scientific affairs. At the same time, he took an active part in the organization of the Polish Academy of Sciences.

The Chair of Electro-acoustics, which was headed by Prof. Malecki for almost twenty years, educated several generations of acousticians and electro-acousticians in this period, mainly in the fields of movie and broadcasting techniques, room acoustics, and design of electro-acoustical equipment. It was the only institution of this speciality in the Warsaw area. The faculty closely cooperated with the industry and polish broadcasting W. Straszewicz, M. Abramczyk, S. Basiński and W. Lenczewski were the first staff members of the chair.

The field of scientific activity of Prof. Malecki in those days was the resultant of the demands of the country in times of reconstruction and industrialization, and of his own scientific interests. Prof. Malecki continued his work on architectural acoustics, which he had started before the war, and presented its synthesis in a book [3].

His designs concerning restoration of the Polish parliament (Sejm), the National Theater in Warsaw, auditoriums, the movie and broadcasting studios, were accomplished. Dr. W. Straszewicz, who was an assistant professor at the Electro-acoustics Chair at those times, was the closest collaborator of Prof. Malecki. He was the author of the acoustical designs of many prestigious halls, such as the that of the National Philharmonic Hall in Warsaw, the Grand Theatre in Łodź, the Philharmonic Hall in Bydgoszcz.

The large-scale reconstruction of the country required the introduction of modern, industrialized building methods, large-panel constructions. That brought new problems for acoustics — design of prefabricated walls and floors, acoustical insulating level of which might be acceptable by the users. In order to attain it, J. Sadowski organized the Laboratory of Building Acoustics at the Building Research Institute, the scientific adviser of which became Prof. Malecki. Development of the measurement methods [4], examination of acoustical properties of the building materials used in Poland, and proposals concerning novel constructions and materials [5–8] were the results of this cooperation.

A few years later Prof. Malecki came back to these problems and he presented, basing on further experimental results of that laboratory and on his own theoretical studies, the theory of impact noises transmission through plate floors [9, 10]. He presented there the application of the theory of wave propagation in an infinite plate to the problem of wave motion in a rectangular plate with boundary conditions simulating the parameters of real floors, the flexural Lamb waves being taking into account. This work attracted the interest of building specialists of the Easteuropean countries of that time.

From the problems of building noise control Prof. Malecki passed to the acoustics of urban planning systems [11], and in this field he co-operated with J. Sadowski.

Looking for more effective methods of industrial noise control, Prof. Malecki started the research on the theory of spatial absorbers, and performed the measurement of a prototype construction [12–15]; this work was conducted in the Chair of Electro-Acoustics.

The pre-war contacts of Prof. Malecki were continued. Optimizing of acoustical conditions in broadcasting studios was still the principal scientific and engineering problem. A considerable part of his books: *The broadcasting and movie acoustics* [16] and *The technique of sound recording* [17] were devoted to that problem. The books were a compendium of the state of knowledge in this field at those times. Original scientific papers of Prof. Malecki, which were connected with this problem, concerned mainly the criteria of acoustical quality of broadcasting studios [18–21]. Prof. I. Malecki presented these papers at several conferences of the Organisation Internationale de Radiodiffusion [22, 23]. This organization accepted the practical conclusions following from those publications.

Apart from the applied research, Prof. Malecki was also interested in the fundamental research, serving as a theoretical basis of different fields of aocustics. In the period discussed, the beginning of the fifties, the foundations of modern acoustics were created [24, 25]; a revision of the classical system of Kelvin's electro-mechano-acoustical analogies became necessary. Firestone formulated an "improved" system, which concerned the lumped constants scheme. Prof. Malecki in several publications [26–31] presented extensions of both the analogy systems to continuous and isotropic media, and also to the transmission lines. An essential aspect of the study consists in treating the acoustic impedance as a wave quantity [32], and in application of the mathematical methods used in telecommunication to the problem of propagation of elastic waves in rods [33]. In his later studies Prof. Malecki frequently used to return to different methods of application of the electro-mechano-acoustical analogies. Many years later he summed up his research in a book form [34] and he kept on dealing with these problems, in cooperation with Dr R. Uklejewski [35].

4. The 1953–1969 period

Let us return to Prof. Malecki's personal history. Working still as a head of the Electro-Acoustics Chair of the Warsaw Technical University, he found a new and a wider field for his activity in the Polish Academy of Sciences (PAN). He became its corresponding member in 1953, and its full member in 1957. In October 1953 the presidium of the Polish Academy of Sciences nominated him for the post of a director of the recently created Institute of Fundamental Technological Research of the Academy. It was (and is) an interdisciplinary institute, and became the largest institute of the Academy a few years later; its most important departements were: mechanics of continuous media, automatics, electronics, and acoustics. Work in the field of acoustics was conducted in the Department for Vibration Research, which was headed by Prof. Malecki, independently of his post of the Director General of the whole Institute. He spent nine years at this post, until he left for the Presidium of the Polish Academy of Sciences in 1962. He acted as a scientific Deputy Secretary of the Academy for more than two tenures.

The Department for Vibration Research was dealing mainly with the ultrasonic methods, what caused a change in the subject matter of Prof. Malecki's scientific research. A team of young, talented acousticians worked in this institution. All of them are professors now. They were, among others,: Prof. J. Wehr, who greatly contributed to the development of measurement methods of the velocity and attenuation of ultrasonic waves [36, 37], and unfortunately died tragically in 1977; Prof. L. Filipczyński, the designer of the first Polish ultrasonic flaw detectors [38], who developed the theory of ultrasonic waves diffraction at obstacles [39] and the electro-acoustic transducers theory. Further on his scientific way, he was most successful in the field of medical ultrasonic diagnostics. The work of the team he headed [40] brought him an international approval. Recently he was elected a member of New York Academy of Sciences. The book written by him, together with Prof. J. Wehr and Prof. Z. Pawłowski [41], was a compendium of knowledge of Polish materials engineers during many years. Prof. Z. PAWŁOWSKI had a great share here as a mechanician and a materials scientist. He also published a series of papers in the field of materials testing methods [42, 43].

Prof. J. KACPROWSKI, the oldest member of this group headed by Prof. Malecki, started his activity in the team creating the foundations of the theory of electro-mechano-acoustic transducers [44]: then he developed the methods of acoustical measurement of noise, e.g. [45], and finally he concentrated on the problems of speech analysis and synthesis, e.g. [46], where he was an unquestionable authority.

In the discussed team, the problems of elimination of noises and vibrations were represented mainly by Prof. S. CZARNECKI. His achievements included the development of the theory of sound control by resonators [47] and screens [48]. He was, together with Prof. Malecki, an initiator of many research and engineering projects, which were undertaken by the team. Unfortunately, he died untimely in his most creative years.

Prof. J. RANACHOWSKI was one of the first co-workers of Prof. Malecki in the Department for Vibration Research, and then in the Institute, where he also acted as the deputy director for many years. Prof. Ranachowski organized a team dealing in a comprehensive way with acoustic properties of brittle materials, especially of ceramics. Starting with his doctor's thesis [145], he developed a theory, based on experimental verification, of dependence of the ultrasonic wave velocity and attenuation on the structure and strength of ceramic materials [49 – 51, 151]. A very high measurement precision was obtained and new general rules were discovered. Results of the research found numerous applications, mainly to testing the high-tension insulators [148, 149].

The problem of reconstruction and development of the Polish mining industry, which was urgent in 1950 – 1955, was an impulse for Prof. Malecki to undertake, together with Prof. W. Kołtoński, the laboratory and *in situ* investigation on application of ultrasound as a tool for prospection of geological strata [52 – 55]. The managerial duties did not allow Prof. Malecki to continue the *in situ* work, which were then taken over by Prof. KOŁTOŃSKI [56] together with Dr. A. JAROSZEWSKA [57]. Prof. Malecki dealt mainly with directing the ultrasonic engineering development in the institute and in the whole country. A series of his synthetical papers [58 – 68] date from this period. They were of a great significance for the development of the Polish acoustics, since they pointed at the research problems which had not been solved yet, suggested priorities, and presented the industrial and social importance of acoustics.

The research on the acoustic properties of ceramics, conducted by the author of this publication, also undertaking the problems of prospecting of geological strata, and next the general aspects of ultrasonic non-destructive testing, directed attention of Prof. Malecki to the theory of propagation of ultrasonic waves in heterogeneous media. The theory has not been entirely developed till now. He focussed his interest mainly on the "granular" media, as they are vastly represented in modern technology [69 – 75]. The idea was based on treating the heterogeneities which appear in a material, e.g. spherical pores in technical porcelain, as a system of spatial sources, which radiate a wave of disturbance. A single obstacle impedance was also considered

[76]. The theory of granular media was verified experimentally and improved theoretically [77] by Prof. J. Ranachowski who used an example of porous ceramic materials [78]. In addition to electrical porcelain, sintered copper materials were tested, which also exhibit porosity, subject to theoretical determination [147].

In the period of his work in the management of the Polish Academy of Sciences in 1962–1969, Prof. I. Malecki was unable to conduct experimental work; however, he kept on working theoretically and, first of all, he completed his monumental monograph — about 700 pages — which was published in Polish in 1964 [79], and then, in an improved and supplemented version, edited in English by Pergamon Press (Oxford) in 1969 [80]. For Polish acousticians it was a basic university handbook and a valuable aid for research work. Unfortunately, both editions are not available nowadays.

In that time the scientific interests of Prof. Malecki were directed to higher frequencies of acoustic waves, in the range of hundreds of megahertz [81, 82].

A transducer utilizing the coupling of the mechano-acoustic field with the electro-magnetic field seemed to be the most promising source of high frequency elastic waves. In theoretical studies, it led to a more general problem of coupled fields theory. Prof. Malecki cooperated here closely with Prof. S. Kaliski, who was a leading specialist in this field at those times. The problem was based on computing the influence of electric or magnetic field on the attenuation and velocity of propagation of the Rayleigh wave in a boundary layer between the media. The wave velocity dispersion effect appears particularly distinct when one of the media is an ideal conductor.

The statement that a feedback appears at a coupling of an electron beam propagating across the boundary surface of two media with an acoustic wave, was of importance for further theoretical research and applications. A negative attenuation of acoustic wave appears when specified conditions concerning the media and the electron beam are fulfilled, so an electronic amplifier appears [83]. Laboratory tests of such amplifiers were conducted by Prof. Kaliski.

The theory of coupled fields was the subject of Prof. Malecki's plenary lecture at the 5th Congress of the International Commission on Acoustics (ICA) in Liége in 1965 [84]. The lecture was given a very good reception by the congress participants.

Attention has been also paid to the necessity of evaluation of nonlinear effects during propagation of acoustic waves. The contribution of Prof. Malecki to this problem was presented in a few papers [85–88], where he drew attention to the significance of the third harmonic in distortion measurement.

5. The 1969–1983 period

The next logical step in Prof. Malecki's scientific interests was transition to the research on the phenomena in the gigahertz band (hypersounds). These are mainly theoretical studies, as they were done in the period when Prof. Malecki was holding important managerial positions — in 1969–1972 he was the director of the

Departement of Science Policy of UNESCO in Paris; after he return home he once again took over the post of the Director of the Institute of Fundamental Technological Research.

Prof. Malecki was fascinated by the question of passage from methods of classical acoustic to the quantum acoustics; he dedicated nearly ten years of his scientific activity to the problem. He published two monographs [89], [90] on this topic, which included original theoretical solutions. He also published a series of articles [91–99].

Prof. I. Malecki's theoretical work on quantum acoustics deals with two slightly different questions: (1) improvement of the quantum acoustic methods for description of systems in which the quantum structure of acoustic field is not approximable by the methods of classical acoustics (it refers mainly to hypersounds); (2) utilization of the quantum method as a convenient tool for analysis of the acoustic field of lower frequencies (including audio-acoustical problems).

In the first scope, the most important results of the research, obtained partially in cooperation with Dr. M.M. Dobrzański [92, 94, 95, 100], included: (a) calculation of phonon scattering by reversion of obstacle spins [94]; (b) application of the Airy equation to the description of boundary conditions in quantum closed system [91, 93]; (c) determination of the range of applications of the Hamilton-Jacobi equation for the boundary conditions [95]; (d) description of interaction of phonons with exitons [99].

Co-operation with Dr. M.M. Dobrzyński lead to results of a large significance. Prof. I. Malecki proposed a formulation of an acoustic wave as a stream of quasi-phonons, which correspond to moving oscillators with distinguished wave numbers [91, 96, 103]. The usefulness of such a presentation was demonstrated on an example of calculation of acoustical scattering around an elastic obstacle in a liquid medium [92].

A new idea was the application of the quantum acoustics method to the description of transmission of sound signals in the internal ear. It has been suggested that the quantum approach is useful for presentation of discontinuous processes occurring during excitation of the hearing organ near the auditory threshold [102, 104, 107, 108]. The auditory canal was presented as a chain of quasi-quantic wave oscillators. The limiting value of the sound intensity incident at the eardrum, at which the application of quantum methods is necessary, has been defined.

The research in the field of quantum acoustics, which has been initiated by Prof. I. Malecki, is developed theoretically and experimentally by his close collaborators. The quantum phenomena are most easily observed in hypersonic frequencies, so the design of resonant and thermal sources of hypersounds by M. Aleksiejuk have been the starting-point. He worked in a team with W. Larecki [105] and S. Piekarski [110], who were dealing with the theory of quantum acoustic fields.

Nowadays, experimental and theoretical studies are going in the direction of testing the acoustical properties of high-temperature superconductors [111] and generation of hypersounds by superconductor junctions [111].

With reference to his earlier works, Prof. Malecki dealt also with developing the electro-mechanical analogies for quantum systems [93, 97].

6. The period from 1983

After his retirement, Prof. Malecki has not reduced his scientific activity; on the contrary, having more time for himself, he could resume the experimental research. Prof. Malecki is still employed in the Institute of Fundamental Technological Research, where he closely cooperates with the team of Prof. J. Ranachowski. This activity resulted in a few synthetical studies [113–118, 119], and in several invited lectures at the 6th Congress of the Federation of Acoustical Societies of Europe (in Zurich in July 1992) [120], and at the 14th Congress of the International Commission for Acoustics (ICA) in Beijing, China, in September 1992 [121].

Prof. Ranachowski's team has undertaken a comprehensive research of the acoustic emission (AE) phenomenon and its application. A modern method of AE measurement has been elaborated. The AE analyzers which have been produced on the basis of the method are widely used by Polish scientific institutions. The research has also allowed to determine, more accurately than before, the dependence of the AE activity on the brittle fracture process, what has opened new possibilities for evaluation by the AE method the strength, "time of life" and fatigue processes of materials [125, 126]. Presently, an investigation of the correlation of the AE activity with electric effects is carried out.

Prof. RANACHOWSKI'S team dealt also with the photo-acoustic method of materials evaluation in collaboration with Prof. I. Malecki [151, 152].

The closest collaborators of Prof. Malecki are: Dr. J. RZESZOTARSKA, who deals with the acoustical emission phenomena in chemical reactions; she has studied a specific acoustic emission effects which occur during the oscillatory reactions [122]; and Dr. Z. RANACHOWSKI, the designer of the analyzer of the acoustic emission effects, who deals with electronic processing of these signals, particularly with application to concrete [123]. He has started to deal, together with Dr. M. MEISSNER and under guidance of Prof. Malecki, with the simulated sources of acoustic emission. The subject of their work [124] consists in designing of the simulated sources and propagation of the signals emitted by them.

The research on the influence of thermo-mechanical processes on the acoustic emission (AE) activity, which was undertaken by the team of Prof. Ranachowski, has been utilized for a comparative study [127] of AE activity during temperature changes in different materials and processes. It enabled the observation of some regularities, based mainly on the asymmetry of the AE activity during heating and cooling of the material. Dr. WITOS, co-operating with Prof. I. Malecki, dealt with the evaluation of utility of the descriptors of AE signals [128]. The research on the AE includes also the theoretical study of Prof. I. Malecki on the determination of the frequency and the amplitude distributions of the AE signals depending on the degree of spatial distribution and correlation of the sources [129].

Recapitulation of previous achievements has been presented in the comparative analysis of the AE applications [130].

7. Organizing activity in Poland

Shortly after the end of the war, Prof. Malecki tried to gather the remaining Polish acousticians and to attract young engineers and scientists to the work on acoustics. At the same time, being aware of the weakness of the Polish acoustic community in those days, he invited several leading foreign acousticians to conferences, organized by him. During the period 1953 – 1966 six such conferences took place, organized by the Institute of Fundamental Technological Research. The first three conferences [131 – 133] were of particular importance due to their pioneering character. They took place in turn at Krynica (1953), at Międzyzdroje (1956), and again at Krynica (1958). They were devoted to ultrasounds and to electro-acoustical transducers. The other conferences, dealing with different sections of acoustics, took place in Warsaw. These conferences were of a great importance for the stimulation of research on acoustics in Poland. They had also a definite international extent, because Prof. Malecki, thanks to his wide personal contacts, managed to assure the participation of prominent acousticians from the countries of Western Europe and from the Soviet Union. During the cold war period it was one of very few opportunities to meet for the two groups of scientists of the same speciality, who hardly knew anything about each other. Advantages of such meetings were evident and fully appreciated by the participants of those meetings.

Scientific cooperation with France was particularly successful. Prof. Malecki and Prof. Pimonow (CNET Paris) were co-chairmen of the Polish – French Colloque sur Ultrasons [134 – 136]. These events took place every two years, alternately in Jabłonna near Warsaw and in Paris, in the years 1978, 1980, 1982, 1984 and 1987. It was a rare example of a systematic co-operation of friendly scientists from both the countries.

The greatest international meeting of acousticians in Poland was the 2nd Congress of the Federation of Acoustical Societies of Europe (FASE), which took place in Warsaw in 1978 [137]. It was the initial period of the activity of this organization — the first congress took place in Paris in 1975. Prof. I. Malecki was the president of the congress. Prof. S. Czarnecki was the president of the Organizing Committee and the leading debate animator. For the first time so many foreign acousticians (about 350 persons took part in the Congress) could acquaint with the Polish achievements in the field.

The conferences were good opportunities for casual meetings; a systematic co-operation of the whole community was also necessary. Apart from the Polish Acoustical Society, which was founded in 1961 by Prof. M. Kwiek (who died in a tragic accident a few years later), there was a necessity of organizing an official representation of the Polish acoustics, and publishing a journal dedicated to acoustics. Prof. Malecki has been devoted to this idea during his whole life: integration of the Polish acoustic community, collaboration, and presentation of achievements independently of the specialities of individual acousticians. Prof. Malecki took up this initiative and, after many efforts, he succeeded in putting into effect the resolution of the Presidium of the Polish Academy of Science, establishing the Committee for Acoustics of the Academy in 1964. Prof. Malecki was appointed the first chairman of this Committee and was being relected to this post for the two next terms until 1969, when he left for Paris. The committee played an important role in presenting the Polish achievements in acoustics to the authorities, to the society, and to other countries. Prof. L. Filipczyński has been the chairman of the Committee since 1970, while Prof. I. Malecki was elected the Honorary Chairman of the Committee.

The quarterly Archiwum Akustyki, which began to appear in 1966, (originally edited by Prof. S. Czarnecki), is being published in English as Archives of Acoustics since 1969 Prof. J. Lewandowski is now editor-in-chief. Prof. I. Malecki was the Chairman of the Editorial Board for many years.

Great importance of co-ordination of the scientific research in the field of acoustics [153, 154] by means of the governmental projects should be also mentioned. These projects contributed, to a great extent, to the animation and thematical integration of the research in Poland.

8. The international co-operation

Prof. I. Malecki is known among scientists of many countries not only as an acoustician but also as one of the initiators of a new scientific field — the science of science. It has strengthened his position on the international ground and, thanks to that, he was elected a vice-president of the International Council of Scientific Unions (ICSU) in 1962, and he held this post until 1966. ICSU is the governing body of sixteen international unions in the field of exact and natural sciences, among others the International Union of Pure and Applied Physics (IUPAP). The International Commission for Acoustics (ICA) is the member of this union. Appreciating the achievements of Prof. I. Malecki in the international field, and his personal output in acoustics, IUPAP designated him to the post of the ICA chairman in 1966 and repeated this election in 1969. He was the president of two ICA world congresses in Tokyo in 1968, and in Budapest in 1971, where he delivered the opening lectures [138] [139] [140], and presented the development of the world acoustics and its future trends. The ICA members from Poland were later Prof. L. Filipczyński (1974 – 80) and Prof. A. Śliwiński (1981 – 89).

The development of acoustics in Europe made it necessary to organize international congresses and symposia not only world wide but also on a regional scale. This is why several prominent acousticians (J. Frenkiel, W. Furrer, H. Zwicker, I. Malecki) initiated the Steering Committee of the Federation of Acoustical Societies of Europe (FASE). The formal foundation of the organization took place at its first congress in Paris in 1975. Since then Prof. I. Malecki has actively taken part in the FASE activities. In 1979 he was elected to the post of the vice-president of FASE. After his term in the office was over, the participants of the FASE meeting in Goettingen elected him a Honorary Member and a life-member of the FASE Scientific Board.

Prof. Malecki is also in a frequent contact with acoustical societies of many countries. He has been elected to a Honorary Member of: the Acoustical Society of Poland, the Acoustical Society of Spain, the Ultrasonic Society of India, the Latin

American Acoustical Society. He is a fellow of the American Acoustical Society. He is a Doctor Honoris Causa of the Technical University of Budapest and of the Academy of Mining and Metallurgy of Cracow.

The appointment of Prof. Malecki to the chairman of the structured session on the Acoustic Emission (AE) at the 14th ICA Congress in Beijing in 1992 [141] was a mark of appreciation for Prof. Malecki's latest scientific works in the field of AE.

9. Education of acousticians

As mentioned above, Prof. Malecki started to lecture at the Warsaw Technical University during the war period. Then, as he was the head of the Electro-Acoustics Chair for twenty years, he had a vast influence on the creation of study programmes in the field of acoustics at this university. Lecturing on fundamental acoustics, architectural acoustics, movie and broadcasting acoustics and ultrasonic engineering, he educated many generations of Polish acousticians. He promoted dozens of dissertations. His lectures on basic acoustics for the first years of study at the Telecommunication Faculty of the Warsaw Technical University contributed to popularization of acoustics among future engineering of other specialities; Prof. Malecki presented his remarks concerning the problems of acoustical education in publications [142, 143].

The integration of the acoustical community and, as a result, creation of the Polish school of acoustics, seems to be the most valuable achievement of Prof. Malecki. The enclosed list of twenty-four doctor's theses, which were promoted by Prof. Malecki, shows the weight of this school. Twelve of Prof. Malecki's postgraduate students are professors nowadays.

The book: *Problems and methods of modern acoustics* [144], which was published on the occasion of Prof. Malecki's 75th birthday, shows the output wealth of the school created by him.

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ACOUSTICS AT THE GDAŃSK TECHNICAL UNIVERSITY

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History of the development of acoustics at the Gdańsk Technical University within the past half-century is briefly outlined. A contribution of the Gdańsk acousticians to the development of acoustics on the country-wide scale is characterized. A decisive influence of Professor Malecki's achievements on the growth of the Gdańsk acoustical milieu is demonstrated.

1. Introduction

Several months ago, on the occasion of the forty years jubilee of the Electronics Faculty at the Gdańsk Technical University an attempt was undertaken to collect data concerning the history of the Faculty, and of the entire Technical University. A jubilee book has been edited on that occasion [5], which contained, among others, many detailed information on the development in acoustics. It is the aim of this article to select those data, and to present them as an entity, together with necessary explanations and appropriate comments.

Starting a historical essay on the development of acoustics one can not refrain from recurring to its most ancient foundations. It is worth mentioning that, although acoustical problems were already treated abundantly by famous philosophers of the antiquity, Phytagoreas, Aristotle, Euclide, or by its engineers such as Vitruvius, and by many others, the name of acoustics appeared but in the seventeenth century, coined probably by Kircher, in his treaty "Fonurgia". Later on, acoustics developed, as a section of physics, especially fast in the eighteenth and nineteenth century, thanks to contributions of a pleiad of renown scientists, from Durerney and Chladni through Young, Fresnel, Fourier, Poisson and Laplace, to Corti, Helmholtz and Strutt-Rayleigh [2].

At the threshold of the twentieth century, electrical and electronic methods applied to acoustics caused an accelerated progress of that interdisciplinary domain. Gradually, electroacoustics, room acoustics, hydroacoustics, aeroacoustics, geoacoustics, audiology, psychoacoustics, sonochemistry, noise control, ultrasonics, molecular acoustics, bioacoustics, sound engineering, musical acoustics, speech recognition and synthesis, sound reinforcement and many other branches of acoustics developed into almost independent disciplines.

A landslide development in the area of acoustic applications has begun with the advent of computer technology. Sound studio techniques applied to broadcasting, film and television, and other media are omnipresent factors influencing everybody's life now. Voice controlled systems and especially voice-operated, speaker-identifying computers will soon become a crucial step in a further revolutionary progress of human civilization.

In spite of such brilliant achievements in the field of applications, the role of acoustics among university disciplines is underestimated. Acoustics has not reached, so far, a faculty level at any university. Laboratories, chairs, or even institutes of acoustics are components of various faculties, either of mechanics, or electronics, or physics, etc. Thus, didactic programs for acoustical studies can not be rationally conceived, being rather a compromise between discrepant directions of teaching. Due to that situation, students graduated in acoustics are too narrowly educated within the fundamental of acoustics. The situation can improve only when acoustics at universities develops into independent faculties.

Creating faculties of acoustics requires, first of all, a formation of their adequate scientific and didactic staff. Before this is achieved, it is worth observing such a development on the example of the Gdańsk Technical University. The knowledge of the history of a development helps in its further progress.

2. Early rudiments

The history of the Gdańsk Technical University began in the year 1904, when it was opened under the name of Technische Hochschule Danzig [4]. Searching for rudiments of acoustics in Gdańsk should concern the periods preceding the first, and the second World War. Helas, documents from those periods were lost during destructions in 1945. However, some data were recovered.

It is namely known that in the year 1908/09, a new organized Chair of Light Technics and Telecommunication conducted laboratories on telecommunication measurements, which had to include some acoustic measurements. Laboratory equipment was founded by the renown enterprise Siemens & Halske. Other facilities were built in the period between the two wars.

In 1945, immediately after the cease of hostilities in Gdańsk, a Polish crew restoring to order the University buildings found an acoustic laboratory in an almost undamaged state. An anechoic chamber in the Building of Electrotechnics was equipped, among others, to particle velocity measurements by the Rayleigh disc method. That laboratory provided a rudimental base of the Chair of Applied Electrotechnics and Acoustics, then organized under the leadership of Professor Ignacy Malecki, at the Electrical Faculty. The Chair started research and teaching in the domain of acoustics, concentrated mainly on problems in architectural acoustics and in hydroacoustics. A former laboratory of the German navy, left in the harbour of Gdynia, was employed for the latter group of problems [3].

3. First achievements

One of the two earliest Ph. doctor's dissertations acquired at the Electrical Faculty, in 1950, concerned hydroacoustics problems. The doctor's degree has been obtained by dr Z. Góra-Zubalewicz, whose promotor was Professor MALECKI [5].

Starting from 1946, lectures on Electroacoustics were given by Prof. Malecki to students of the third year, at the sections of teletechnics and radiotechnics. The lectures continued till 1951, when Professor Malecki was transferred to Warsaw [5]. Earlier, however, he wrote a series of Polish original textbooks on acoustics, very important for scientific and didactic purposes in a recently organized university.

The first book, entitled "Mechanism of the sound propagation in rooms", was edited in 1949 by a publishing house A. Krawczyński, in Gdańsk, and sponsored by a Gdansk students' editing commission "Bratnia Pomoc" [1]. The book was an enlarged version of the second dissertation (habilitation), written by Prof. Malecki during the war period, in the years of the heaviest Nazi terror in Poland. Scientific activities in that period, as it is said in the book foreword, became fighting tools for the liberation of the country. The text was preceded by a review of late Professor M. Wolfke, the eminent Polish physicist, who highly appreciated the value of the book, emphasizing its role as the first Polish scientific publication on room acoustics.

4. A pause

Professor Malecki's transfer to Warsaw caused a pause in acoustical activities at the Gdańsk Technical University. However, the pause was not absolute. Professor Malecki acted as promotor or reviewer of dissertations in the domain of acoustics. Acoustical problems entered in research works and didactic programs of various chairs at the Faculty of Electronics (earlier Faculty of Communication). Among others, the Chairs of Radionavigation, of Radiocommunication, of Fundamentals of Telecommunication, and partly others, applied to the studies of acoustical problems. Some studies connected to acoustics were also pursued at the Mechanical Faculties. Those activities prepared a necessary basis, first of all, a qualified staff of academic teachers ready to start again a full scale scientific progress in the domain of acoustics.

5. Continuation of achievements

During a reorganization of the Electronics Faculty, in 1969, a department was created, destined exclusively to acoustical research and teaching. Organized by Professor Zenon Jagodziński, under the name of Hydroacoustics and Electrophony Department, it comprised two separate Laboratories: Hydroacoustic one and Electrophonic one. Later on, in 1982, the Laboratories developed into independent Departments under the names of Hydroacoustics, and of Sound Engineering. Both Departments underwent a fast, continuous process of development, becoming a strong scientific centre, which helped in research and education in other institutions.

The development process may be characterized, on one hand, by a number of graduations obtained in acoustics at the both mentioned departments. The total number exceeded five hundreds MSc and BSc degrees. On the other hand, the professional level of the graduates is highly assessed by institutions where they work, many of them working abroad at very advanced positions.

A number of dissertations in acoustics obtained at the Electronics Faculty TUG is also an attribute to the achievements of the Gdańsk acousticians. The total number of 26 promotions, among them 19 titles of doctor of science and 7 of habilitated doctor of science, including those obtained at other universities by this Faculty scientists, were awarded so far [5].

Recently, after a reorganization of the Faculty, a new unit has been created, namely the Chair of Acoustics, which includes the Departments of Sound Engineering and of Hydroacoustics, as well as some specified laboratories.

6. Acoustics at other Faculties of the Technical University

Besides of the Electronics Faculty, acoustics problems were practised at some other faculties of the Technical University, in cooperation with acousticians-lecturers from Electronics: e.g. at Architecture, where lectures on Architectural Acoustics and Environmental Acoustics were given, or at Mechanics, where research cooperation in acoustic measurements techniques were maintained, or Shipbuilding, where underwater acoustics problem were of common interest. Series of lectures on environmental acoustical problems were also given to all students within a general education course.

7. Acoustics at other Gdańsk educational institutions

Acoustics at the Electronics Faculty also influenced the developments of activities outside the Technical University. Both acoustical Departments cooperated and helped actively in the creation of the Environmental Laboratory of Acoustics, organized at the Gdańsk University. At the Gdansk Academy of Music, a Laboratory of Musical Acoustics was created, as well as courses on Technology of Experimental Music were introduced there, thanks to the participation of the TU Sound Engineering Department.

A long term scientific and didactic cooperation was established between the Department of Hydroacoustics and the Naval Academy in Gdynia. Several other departments of the Faculty, as well as of the former Shipbuilding Faculty, actually Oceanotechnics Institute, participated in that cooperation.

Cooperation with local industry plants, as well as with those all over the country, concerning acoustical problems was also developed. Especially worth mentioning was a close cooperation with the Polish Committee on Radio and Television, which comprised not only didactic, but also scientific and productional aims.

The cooperation started in the fifties, when an educational formation of the Polish Radio technical staff in Northern Poland was entrusted to acousticians of the TUG Electronics Faculty, under the supervision of M. Sankiewicz. More than hundred of proficiency certificates and titles of technicians have been conferred during that period of theoretical and practical training. Further cooperation developed into common scientific and production activities. A commonly organized stereophonic sound studio at the Electronics Building produced numerous sound recordings for broadcasting through Gdańsk Radio and TV transmitters, as well as for the central program emitted from Warsaw. Those activities became an excellent basis for professional formation of students, graduating in Sound Engineering, and gained a durable popularity of this direction of teaching among Faculty candidates.

8. Gdańsk share in national and international acoustical cooperation

Scientific and, especially, didactic achievements are generally difficult to be valued or appreciated. However, they may be estimated on the ground of presentations made by authors before a competent audience, such as usually gathers at the meetings of scientific societies. Therefore, contributions to society activities may be, to some extent, treated as a criterion of achievement quality on a larger than local scale.

In the sixties, Gdańsk acousticians joined the Polish Acoustical Society, governed then by the Poznań centre, with Professor Helena Ryffert as Society President. In 1981, however, Professor Zenon Jagodziński, then head of the TUG acousticians, was elected the PAS President. His successor, Professor Antoni Śliwiński, the actual PAS President is from Gdańsk University too. The Gdańsk Branch of the PAS organized in the seventies and eighties several country-wide Open Seminar on Acoustics, with the participation of acousticians from abroad. The Gdańsk seminars outnumbered other ones relative to the number of papers and participants.

In the meantime, other specialized symposia were organized by the Gdańsk PAS Branch, first of all, Symposium on Hydroacoustics, held yearly since 1984, and Symposium on Sound Engineering, held biannually since 1985.

An important world-wide acoustical event, and an outstanding success of the Gdańsk acousticians from both the University and the Technical University, was the organization of an international conference entitled "Prospects in Modern Acoustics — Education and Development" held at Jastrzębia Góra near Gdańsk, in 1987. The concept of this conference and its affirmation by the International Commission on

Acoustics, as well as by the International Union of Pure and Applied Physics, was again due to Professor Malecki, while Professor Śliwiński acted as President of the Organizing Committee [6].

Gdańsk acousticians participated more and more frequently in conferences and congresses organized abroad. Besides, they went to training periods or to visiting-professor contracts to several foreign universities or scientific centres, e.g. to France, Great Britain, Germany, Greece, Denmark, United States of America, Canada, etc., being revisited by their foreign partners. The Gdańsk acousticians are members of several foreign acoustical societies, e.g. Société Française d'Acoustique, Acoustic Society of America, Audio Engineering Society, Hellenic Acoustical Society, Deutsche Arbeitsgemeinschaft für Akustik, etc. Experience gained from all the above mentionned activities permitted to undertake a new important initiative.

In 1991, Gdańsk acousticians have been authorized by the Governing Bodies of the Audio Engineering Society to create a new regional section of the AES. Then, the Polish AES Section was founded in Gdansk, which was become the Board site of the new scientific society in Poland. Actually the Polish AES Section, under the chairmanship of M. Sankiewicz, has already about one hundred members, which actively participate in society actions.

9. Concluding remarks

A short essay on the development of acoustics in the Gdańsk Technical University does not permit all achievements, to be properly characterized and presented with an appropriate balance. Their value might be comparatively estimated if faced with the achievements of other scientific centres in Poland, or, moreover, to those abroad. Such estimation would need enlarged studies, which actually are beyond possibilities of the present authors.

However, in order to deliver another information, more appropriate for comparisons, a total number of publications written by the acousticians of the Technical University of Gdańsk has been evaluated, basing on recently published data [5]. This number amounts to 931 publications, including co-authors' items, written by 51 acousticians. At any rate, it seems to be an important contribution to the developments of acoustics in Poland.

A final conclusion occurs irresistibly. The achievements done by Professor Malecki during the period of his activity in Gdańsk, turned out to be fruitful, despite the period of a supposed pause. Thus, the Gdańsk Technical University has become an important scientific centre for Acoustics, especially for Hydroacoustics and for Sound Engineering, i.e. exactly those two acoustical domains initiated then by Professor Malecki.

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CATEGORICAL PERCEPTION IN ABSOLUTE PITCH

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Three music students, possessing of absolute pitch, participated in an experiment of categorical identification of two adjacent musical pitch categories A$\#_4$ and B_4. Next, they also participated in the experiment in which their ability to differenciate between two tone pulses separated by frequency level distance of 25 cents was tested. When ABX procedure was used with the tone-pulse length reduced to 20 msec and the interstimulus interval extended to 10 sec, the between-category discrimination in two sucjects markedly exceeded the within-category discrimination, which signalled the existence of categorical perception.

1. Introduction

The words "absolute pitch" (AP) mean the ability of some musically trained people to recognize musical tones of a desired pitch (passive AP) or both to recognize and produce them (active AP) without being given any tone of reference (Bachem [1], Rakowski and Morawska-Bungeler [15]). Most poeple, also including professional musicians, do not possess this ability, which possibly can be developed only in early childhood (Ward [16]). Instead, they can practice and develop the ability to recognize and to produce a number of frequency-ratio categories called musical intervals. The ability to deal with musical intervals and their sequences (melodies) is called "relative pitch" (RP).

The phenomenon of absolute pitch can be described as the existence in the long-term memory of a set of 12 standards and corresponding "chromas" or categories. These standards, which are imprinted in the long-term memory with considerable accuracy, serve in the formation of those categories as salient points along the frequency-ratio continuum (Morawska-Bungeler and Rakowski [12]). The chroma categories are nearly a semitone wide and in "good quality" AP possessors have sharp, well-defined boundaries.

The shapes of chroma-category boundaries are very similar to those obtained in experiments concerning the perception of synthetic speech sounds. In course of such experiments performed at Haskins Laboratories, Liberman et al. [7] found and described a phenomenon called "categorical perception". Categorical perception means sharp and well-defined boundaries between adjacent perceptual categories and systematic differences in possibility to discriminate sounds. Differences depend on whether the two sounds being compared belong to the same category or to different categories.

A large number of experiments confirmed the above-described property of categorical perception in speech sounds. A study by Liberman and his colleagues from Haskins Laboratories may serve as an example (Liberman et al. [8]). They produced synthetic speech stimuli which varied in acoustically equal steps through the range sufficient to produce the initial stop consonants /d/ and /t/. Such stimuli were perceived as members of two discrete categories separated by a relatively sharp boundary. Additionally, when listeners heard adjacent pairs of these stimuli they could either easily discriminate between them or could not find any difference. The condition under which the adjacent stimuli were perceptually different was that they were taken from both sides of the boundary separating phonemic categories. When they were both taken from the same category, subjects could not discriminate between them, though the physical distance which separated both stimuli was in both cases exactly the same.

The theory that underlies the above-described phenomenon says that in the speech-perception mode people can discriminate among speech sounds only in as much as they can identify them. This statement lies within the context of the motor theory of speech perception (Liberman et al. [9]); categorical perception is assumed to result from the categorical nature of speech production. Different sounds are produced and perceived under the same articulatory set of commands.

The ideal case of categorical perception under the assumptions of the above-mentioned theory is shown in Figure 1. Eight stimuli are spaced at equal intervals along a physical continuum. The stimuli 1 – 4 are identified as members of Category A, stimuli 5 – 8 as members of Category B. The identifications are completely consistent and boundaries between the categories are sharp. When stimuli 1 – 8 are successively presented subjects hear no change between stimuli 1 – 2, 2 – 3, 3 – 4, and then perceive an abrupt change between stimuli 4 and 5, as category changes from A to B. Adjacent stimuli in the range 5 – 8 are again perceived as identical. If the discrimination is measured, e.g. by an ABX procedure, performance is at chance level for all pairs of adjacent stimuli except for the pair 4 – 5 where it is perfect.

The early adherents of categorical perception strongly insisted that the phenomenon was a unique feature of speech perception. However, it soon became clear that to some degree it may be observed in experiments with non-speech stimuli and even with non-human listeners. Miller et al. [11] found categorical perception using stimuli constructed of a wide-band noise and a low-pitched buzz. The buzz started with varying time delay after the noise onset; this simulated the voice onset time (VOT) in various stop consonants. Kuhl [5] found categorical perception in exploring chinchillas' ability to diferentially "label" stimuli with various VOT. Locke and Kellar [10] uncovered evidence of categorical perception in trained musicians judging triadic

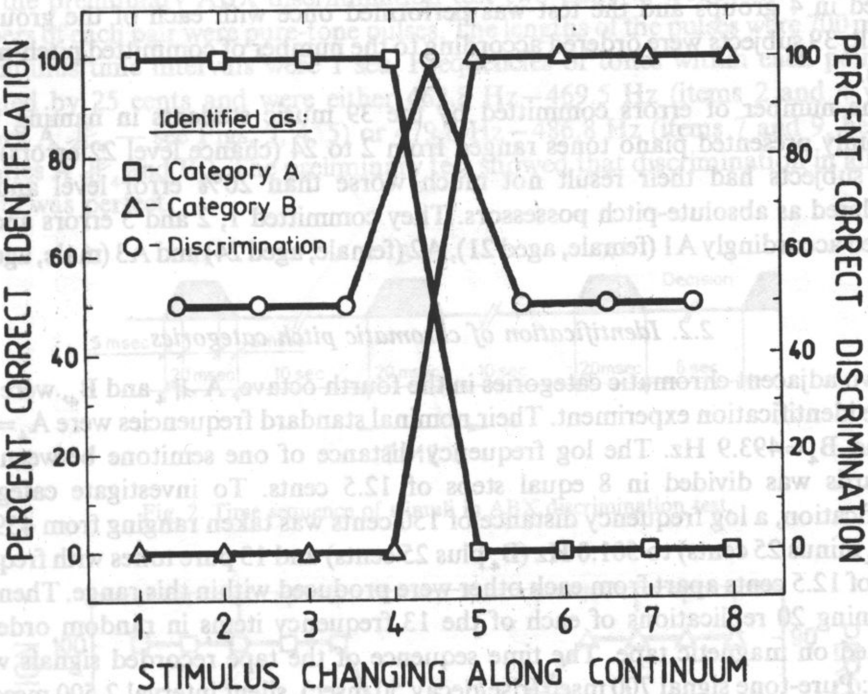

Fig. 1. Idealized identification of stimuli belonging to two categories and corresponding discrimination of adjacent stimuli according to categorical perception.

chords whose middle note varied. Burns and Ward [2] studied categorical perception of melodic musical intervals, and Clarke [3] found it in listening to musical rhythmic patterns. The aim of the present experiment was to check whether elements of categorical perception may be found in the perception of musical pitch categories by the possessors of absolute pitch.

2. Experiments

2.1. Subjects

In order to find subjects who possessed absolute pitch and could participate in the main experiment, the so-called pitch-naming test was applied to a group of 39 music students. The test, which was previously used in testing another group of students (Rakowski and Morawska-Bungeler [15]) consisted in a 24-item series of piano tones taken semi-randomly from the whole range of the instrument. The tones were recorded on tape and presented to the subjects through a loudspeaker and with moderate loudness. Time distances between the onsets of subsequent tones were 4 seconds and the subjects had to write the musical name of each tone (e.g. G, C or D) on an answer sheet. The subjects were divided in 4 groups and the test was performed once with each of the groups. As a result, 39 subjects were ordered according to the number of committed pitch-naming errors.

The number of errors committed by the 39 music students in naming the 24 randomly presented piano tones ranged from 2 to 24 (chance level 22 errors). Only three subjects had their result not much worse than 20% error level and were considered as absolute-pitch possessors. They committed 1, 2 and 5 errors and were labelled accordingly A1 (female, aged 21), A2 (female, aged 24) and A3 (male, aged 24).

2.2. *Identification of chromatic pitch categories*

Two adjacent chromatic categories in the fourth octave, $A\#_4$ and B_4, were taken for the identification experiment. Their nominal standard frequencies were $A_4=466.2$ Hz and $B_4=493.9$ Hz. The log frequency distance of one semitone between these standards was divided in 8 equal steps of 12.5 cents. To investigate categorical identification, a log frequency distance of 150 cents was taken ranging from 459.5 Hz ($A\#_4$ minus 25 cents) to 501.0 Hz (B_4 plus 25 cents) and 13 pure tones with frequency levels of 12.5 cents apart from each other were produced within this range. Then a test containing 20 replications of each of the 13 frequency items in random order was recorded on magnetic tape. The time sequence of the tape recorded signals was as follow: Pure-tone signal 700 msec (rise/decay 50 msec), silent interval 2.500 msec. The whole test lasted nearly 14 minutes and was presented to the subjects with 5 three-minute breaks. Before the testing started, subjects were given some practice in recognizing chroma categories in pure-tone stimuli. Frequencies of tones used in this pre-testing were taken from categories other than A # and B.

The subject's task in the identification test was to respond to each sound stimulus by marking on an answer sheet the name of the category (A # or B) to which the item belonged. The number of the item was signalled visually, and the choice was obligatory. The test was presented from a loudspeaker with a loudness level of 65 phons to each subject individually in a sound insulated room. The results of the identification test are shown as percent correct identification for subjects A1, A2 and A3 in Figs. 3, 4 and 5.

2.3. *Discrimination across and within categories*

In order to check the existence of categorical perception discrimination tests should be performed using pairs of stimuli taken either from the same category or from neighbouring categories across the boundary. The width of the border region between the chromatic categories of our subjects required that a two-step discrimination test should be applied. It meant that the stimuli to be discriminated should have their frequencies 25 cents apart. However at such large frequency differences frequency discrimination is very easy (Rakowski [14], Wier et al. [17]). To check this a preliminary test of frequency discrimination was applied.

In the preliminary ABX discrimination test two pairs of frequencies were used. Members of each pair were pure-tone pulses. The lengths of the pulses were 700 ms and interstimulus time intervals were 1 sec. Frequencies of tones within each pair were separated by 25 cents and were either 462.8 Hz–469.5 Hz (items 2 and 4 within category A $\#_4$ — see Figs. 3, 4, 5) or 479.8 Hz–486.8 Hz (items 7 and 9, between categories A $\#_4$ and B_4). The preliminary test showed that discrimination in all three subjects was perfect.

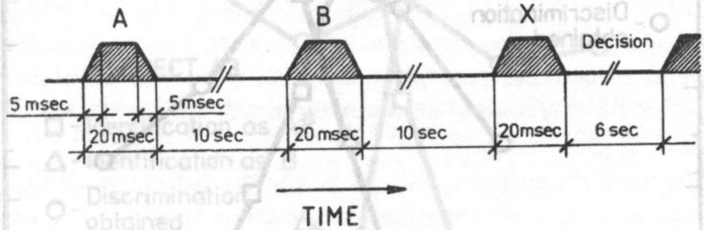

Fig. 2. Time sequence of stimuli in ABX discrimination test.

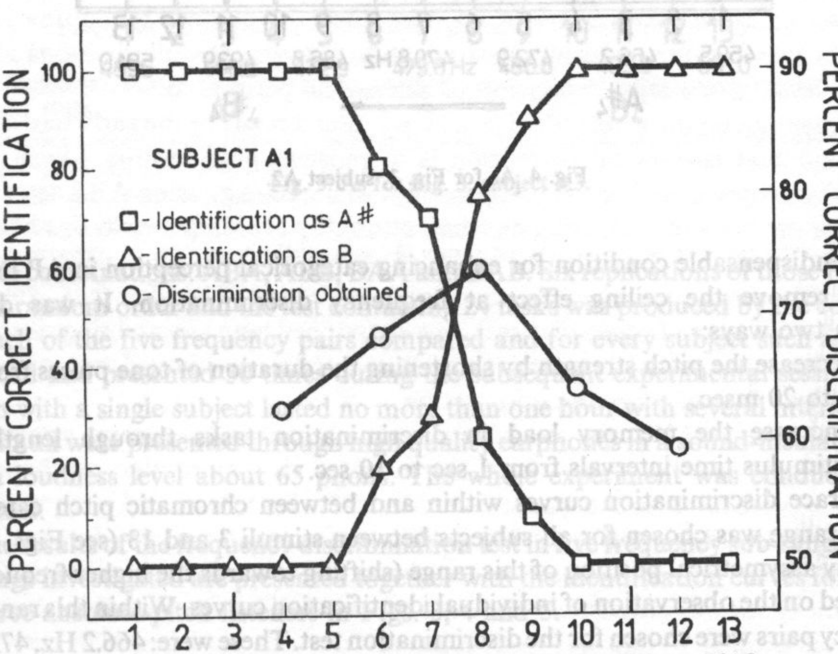

Fig. 3. Categorical identification by subject A1 of thirteen 700-milisecond tone pulses with various frequencies as belonging to pitch category A or B. Discrimination between pairs of 20-milisecond stimuli separated by two frequency steps shown for the same subject (720 decisions per point in ABX test with 10 sec interstimulus intervals).

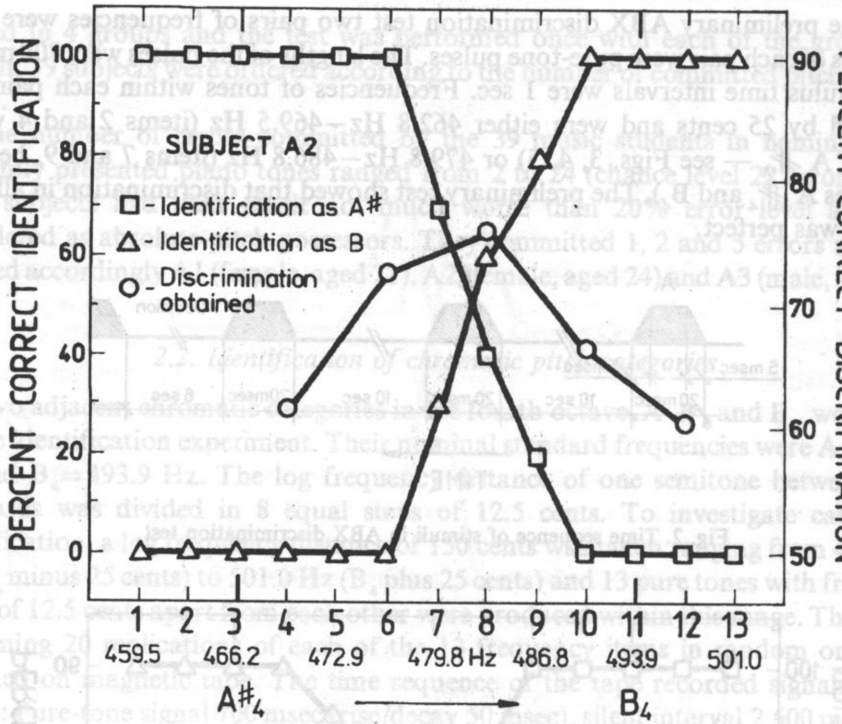

Fig. 4. As for Fig. 3, subject A2.

The indispensable condition for enhancing categorical perception in AP listeners was to remove the ceiling effect at frequency discrimination. It was decided to act in two ways:

1) To decrease the pitch strength by shortening the duration of tone pulses from 700 msec to 20 msec.
2) To increase the memory load in discrimination tasks through lengthening interstimulus time intervals from 1 sec to 10 sec.

To trace discrimination curves within and between chromatic pitch categories a single range was chosen for all subjects between stimuli 3 and 13 (see Figs. 3–5). A slightly assymetrical position of this range (shifting towards the higher frequencies) was based on the observation of individual identification curves. Within this range five frequency pairs were chosen for the discrimination test. These were: 466.2 Hz, 472.9 Hz (stimuli 3 and 5), 472.9 Hz – 479.8 Hz (stimuli 5 and 7), 479.8 Hz – 486.8 Hz (stimuli 7 and 9), 486.8 Hz – 493.9 Hz (stimuli 9 and 11), and 493.9 Hz – 501.0 Hz (stimuli 11 and 13).

The time sequence of stimuli in the discrimination test ABX is shown in Fig. 2. A and B were two tone pulses being compared, and X was one of them (with equal probability). The subject's task was to tell whether X was A or B. There were four

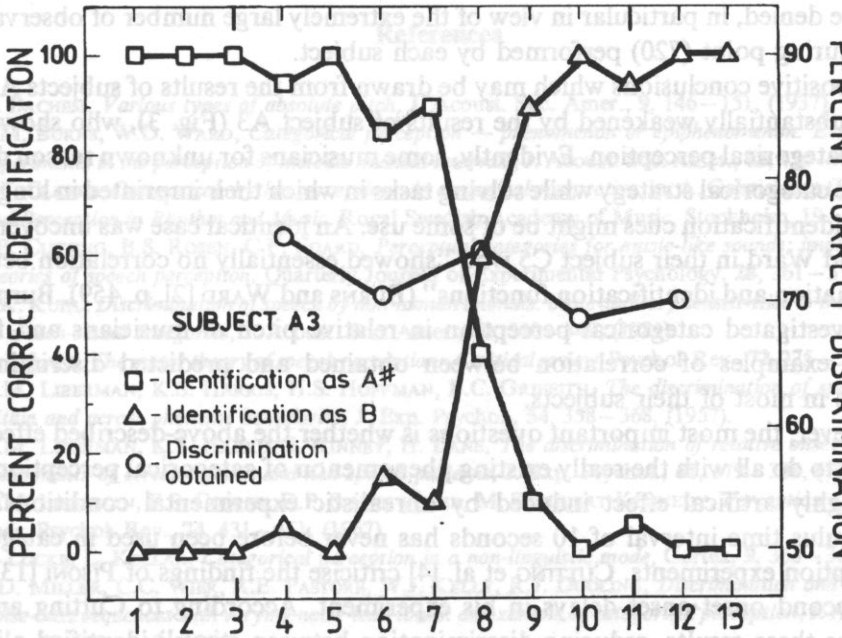

Fig. 5. As for Fig. 3, subject A3.

possible combinations: ABA, ABB, BAA and BAB. Six replications of those four were used in random order and the test containing 24 tasks was produced by the computer. For each of the five frequency pairs compared and for every subject such a test was produced and presented 30 times during the subsequent experimental sessions. One session with a single subject lasted no more than one hour with several intermissions. The stimuli were presented through high quality earphones in a sound-insulated room with a loudness level about 65 phons. The whole experiment was conducted over 3 months.

The results of the frequency discrimination test in five frequency sub-ranges within the range investigated are presented together with the identification curves for each of the three absolute-pitch listeners in Figs. 3, 4 and 5.

3. Discussion and conclusions

As can be seen in Figs. 3 and 4, two subjects, A1 and A2 revealed some enhancement of discriminating ability in the between-category frequency region in comparison with the within-category judgements. The maximum enhancement is only about 15%, nevertheless some degree of categorical perception in subjects A1 and A2 cannot be denied, in particular in view of the extremely large number of observations per measuring point (720) performed by each subject.

The positive conclusions which may be drawn from the results of subjects A1 and A2 are substantially weakened by the results of subject A3 (Fig. 3), who showed no trace of categorical perception. Evidently, some musicians for unknown reason do not adopt the categorical strategy while solving tasks in which their imprinted in long-term memory identification cues might be of some use. An identical case was uncovered by Burns and Ward in their subject C5 who "showed essentially no correlation between discrimination and identification functions" (BURNS and WARD [2], p. 459). Burns and Ward investigated categorical perception in relative pitch of musicians and found excellent examples of correlation between obtained and predicted discrimination functions in most of their subjects.

However, the most important questions is whether the above-described effect has anything to do all with the really existing phenomenon of categorical perception. Is it not a highly artifical effect induced by unrealistic experimental conditions? The interstimulus time interval of 10 seconds has never before been used in categorical — perception experiments. CUTTING et al. [4] criticise the findings of PISONI [13] who used 2-second onset-onset delays in his experiment. According to Cutting and his colleagues these results, reducing discrimination between stimuli identified alike to a chance level, give a false impression of categorical perception.

Commenting on those remarks I would say, that already in the early sixties it had been found that categorical perception is a transient phenomenon which can be induced by properly selecting the measuring technique (e.g. using ABX tests rather than other methods of measuring discrimination), by avoiding over-trained listeners etc. (see e.g. LANE [6]). Nevertheless much effort has been put into investigating this effect in the various domains of human communication. The idea that categorical perception is a unique feature of speech was abandoned long ago and since then it has been considered as one of the efficient strategies that an organism adopts to deal with the abundant flow of incoming information.

It may be argued that the special measures adopted in the present experiments, though strange and "artificial" from the point of view of routine psychoacoustic methodology are far from being unrealistic in real life. Both decreased pitch strength of musical signals and comparing pitch through a ten-second time delay do occur in real concert practice. E.g. such situations may be typical of a conductor listening to the intonation of various instruments during a rehearsal. If he does have absolute pitch, the strategy adopted by him may easily follow the one that was adopted by some of our listeners.

Acknowledgments

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ON THE MODEL OF VOCAL TRACT DYNAMICS

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An attempt to model a true vocal tract shape is presented. The base data were articulatory data taken from the cineradiographic recordings and the speech signal simultaneously registered (Bolla, Földi, 1987). Moreover, information concerning distances between the reference points, labiograms, photopalatograms and photolinguograms were used. A non-uniform three-dimensional vocal tract model was applied. The most difficult problem was how to evaluate the variations of the lateral dimensions along the longitudinal axis of the vocal tract. Only the lip opening dimensions for each speech sound were the objects of direct measurements. After the photo of the subject's hard palate one of the several plaster casts made for other male subjects was chosen as the most similar one. On its basis the dimensions in this region were reconstructed and assume to be constant. The lateral dimensions in other parts of the vocal tract were reconstructed after the formant frequencies measured for each Polish vowel spoken by the subject whose vocal tract was modelled. These dimensions were adjusted in such way that they were common for all vowels and the corresponding formant frequencies were obtained by varying vertical dimensions according to the X-ray data. This model was applied to study the relationship between the shape variation of the vocal tract and its acoustic output, especially in the case of study of transient sounds spoken with vocalic neighbourhood.

1. Introduction

Vocal tract model computation in the frequency domain is nowadays a well established procedure, especially for stationary speech sounds (Fant, 1960; Mrayati, 1978; Atal and al. 1978; Badin and Fant, 1984; Lin, 1990). However, in most cases, the main object of vocal tract simulation is to obtain the best possible match of the calculated frequency responses with characteristics of the modelled speech sounds. Even if the vocal tract configuration taken for calculation is initially based on X-ray picture, it is next modified in order to achieve a higher degree of modelling accuracy in the frequency domain. It must be stressed that the continuous change of the tongue shape and position is hard to measure and to model adequately. Some basic facts are known, which describe certain stable articulatory mechanisms either in the steady state or in transitions; the rest is rather hypothetical. Another source of discrepancy between the vocal tract shape and its physical representation is its approximation by a number of contignuous cylindrical tubes consisting, generally, of about 1 cm long, 17–20 segments. Although, in almost all systems, the vocal tract is represented by a cascade of cylindrical tubes of finite length, the form of the vocal tract cross-sections is far from circular, especially, in its pharyngeal and palatal parts. In the model calculation the cross-sectional shape has a certain influence on surface losses, but their effect on its frequency response is not considerable. The model composed of cylindrical lossy tubes was applied successfully in many researches and was the basis for formulation of the acoustic speech production theory (Fant, 1960).

Even an elementary articulatory model introduces an additional level for representation of speech phenomena such as coarticulation, reduction, assimilation and other context-dependent allophonic variations. This additional level appears to be more suited to human intuition in the manipulation of hypothesis than the lowest level of the exclusively acoustic signal description. Furthermore, new experimental possibilities are open to speech researcher at the acoustic level: some simple articulatory movements may induce very complex acoustic mechanisms that would be not recognized as basic in the speech production process.

However, the uniform cylindrical vocal tract (VT) model applied to reproduce some dynamic phenomena existing in natural speech does not always fit to describe them, especially in the output signal domain. Adopting linear form of the motion from the starting shape to a target one, the output signal calculated for VT of circular cross-section shape reveals stepwise variations non observed in natural signal. This signifies that the interpolated transitional VT configurations have no place in reality. In other words, the variations of the VT configuration (more exactly, of the area function) are not linear and the variations of the cross-section shapes should also be taken into account. This was the reason for applying a non-uniform vocal tract model to shape as precisely as possible, the contours of supraglottal organs obtained from cineradiography.

Application of an articulatory model instead of the physical one offers the advantage that the set of possible area functions is constrained to be compatible with anatomy, thus reducing ambiguities of solutions. In this work we tried to fit our articulatory model to a part of the accessible area function obtained by "direct" methods and fitted it at the same time to the acoustic data. The main criterion used to evaluate the quality of the vocal tract configuration approximation was the degree of discrepancy in formant space between the vocal tract transfer function and the frequency characteristic of the simulated speech sound.

We applied a non-uniform model in order to study dynamic phenomena in speech, in sequence of voiced sounds, like liquids and nasals with vowels, in particular. The main problem was to establish how the vocal tract configuration is changing from one position of the articulators corresponding to a preceding sound to the following one. An important part of the research was devoted to model the steady vocalic vocal tract shape suitable for the natural one in articulatory and formant frequency spaces. An example of mapping from the space of articulators to the frequency space in dynamic case was established for vocalic sequence of /u/ and /i/, with restriction to the time signal to be without jumps.

2. Acoustic model of the speech producing system

The human speech sound is characterized by the properties of the source of excitation and the acoustic transmission system. When assuming purely one-dimensional acoustical wave propagation in the vocal tract, it is known that the most accurate computational model is its representation as a transmission line analog. The vocal tract shape is usually modelled by a series of cylindrical tubes of finite length (Fig. 1).

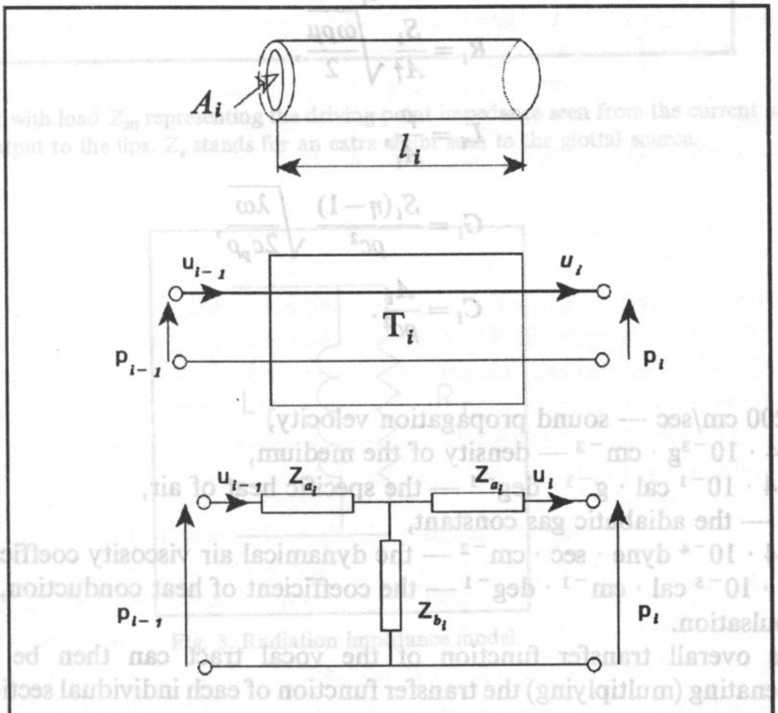

Fig. 1. T-network representation of a cylindrical section of length l_i.

The sound propagation in each cylindrical section is solved by applying the conventional quadrupole equations defining the relations between the input and output quantities (usually acoustic pressures and volume velocities)

$$\gamma_i = \alpha_i + j\beta_i = \sqrt{Z_{a_i} Z_{b_i}} = \sqrt{(R_i + j\omega_i L_i)(G_i + j\omega_i C_i + Y_{w_i})}, \tag{1}$$

with the condition that $|\gamma_i|\, l_i << 1$, and where

$$Z_{0_i} = \sqrt{\frac{Z_{a_i}}{Z_{b_i}}},$$

$$R_i = \frac{S_i}{A_i^2}\sqrt{\frac{\omega\rho\mu}{2}},$$

$$L_i = \frac{\rho}{A_i}, \tag{2}$$

$$G_i = \frac{S_i(\eta - 1)}{\rho c^2}\sqrt{\frac{\lambda\omega}{2c_p\rho}},$$

$$C_i = \frac{A_i}{\rho c^2}.$$

Here
$c = 35200$ cm/sec — sound propagation velocity,
$\rho = 1.14 \cdot 10^{-3}$g $\cdot$ cm^{-3} — density of the medium,
$C_p = 2.4 \cdot 10^{-1}$ cal $\cdot$ g^{-1} $\cdot$ deg^{-1} — the specific heat of air,
$\eta = 1.4$ — the adiabatic gas constant,
$\mu = 1.84 \cdot 10^{-4}$ dyne $\cdot$ sec $\cdot$ cm^{-2} — the dynamical air viscosity coefficient,
$\lambda = 5.5 \cdot 10^{-5}$ cal $\cdot$ cm^{-1} $\cdot$ deg^{-1} — the coefficient of heat conduction,
ω — pulsation.

The overall transfer function of the vocal tract can then be obtained by concatenating (multiplying) the transfer function of each individual section defined as

$$H_i(\omega) = \frac{u_i}{u_{i-1}}. \tag{3}$$

To avoid matrix notation, we should first work out the input impedance looking downstream from the right-hand terminal of the $(i-1)$th section $Z_{B,i} = P_i/u_i$ (see Fig. 2).

The radiation impedance terminates the vocal tract at the front end. An accurate mathematical treatment of this impedance requires to regard it as a vibrating piston (at the lip opening) set in a spherical baffle. But this mode of approximation is rather complex and several simplified radiation models have been proposed for practical applications. One of them (Fant, 1960) has the form presented in Fig. 3.

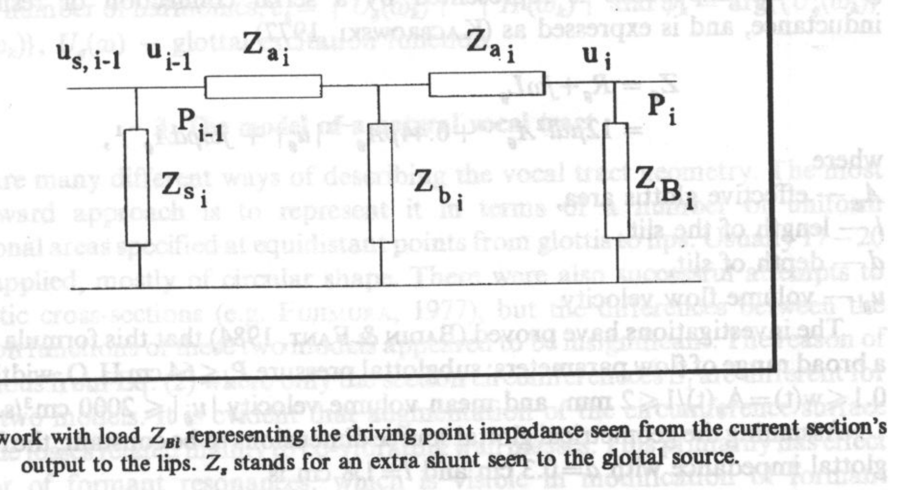

Fig. 2. T-network with load Z_{Bi} representing the driving point impedance seen from the current section's output to the lips. Z_s stands for an extra shunt seen to the glottal source.

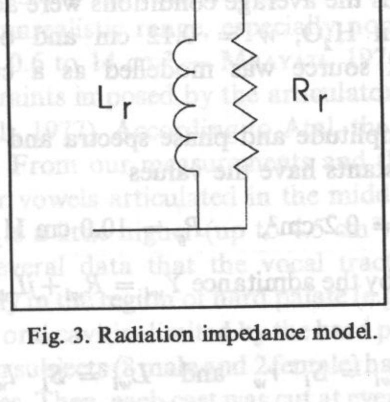

Fig. 3. Radiation impedance model.

Then the radiation impedance is described by following expressions (Wakita and Fant, 1978):

$$Z_r = \frac{K_s \rho \omega^2}{4\pi c} + \frac{j\omega 8\rho}{3\pi\sqrt{\pi A_0}} = R_r + j\omega L_r, \tag{4}$$

with

$$K_s(f) = \begin{cases} 1+(0.6/1600)f & 0 \leqslant f \leqslant 1600 \text{ Hz}, \\ 1.6 & f > 1600 \text{ Hz}. \end{cases} \tag{5}$$

The model of the glottal source applied to calculations is rather simple and no interaction between the voiced source and the vocal tract is assumed. Then, the source of excitation for sonorants is defined as the volume velocity (u_g) of airflow through the glottis. Its impedance is represented by a serial connection of resistance and inductance, and is expressed as (KACPROWSKI, 1977)

$$Z_g = R_g + j\omega L_g$$
$$= 12\mu dl^2 A_g^{-3} + 0.44\rho A_g^{-2}|u_g| + j\omega\rho d A_g^{-1}, \tag{6}$$

where

A_g — effective glottis area,
l — length of the slit,
d — depth of slit,
u_g — volume flow velocity.

The investigations have proved (BADIN & FANT, 1984) that this formula is valid for a broad range of flow parameters: subglottal pressure $P_s \leqslant 64$ cm H_2O, width of the slit $0.1 \leqslant w(t) = A_g(t)/1 \leqslant 2$ mm, and mean volume velocity $|u_g| \leqslant 2000$ cm³/s.

Assuming the geometrical glottal source dimensions to be constant, the formula for glottal impedance with $d = 0.3$ cm and $l = 1.8$ cm is

$$Z_g = 3.72 \cdot 10^{-4} \frac{l}{w^3} + 2.3 \cdot 10^{-2} \frac{1}{w}\sqrt{P_s} + j \cdot 1.95 \cdot 10^{-3} \frac{f}{w}. \tag{7}$$

For impedance calculations the average conditions were accepted, valid for medium voice effort $P_s \approx 10$ cm H_2O, $w = 0.12$ cm and over the frequency range $\leqslant 4500$ Hz. The larynx source was modelled as a current source generating pulses of triangular form.

For calculations of amplitude and phase spectra and the output signals of the modelled sounds, the constants have the values

$$A_g = 0.2 \text{ cm}^2, \qquad P_g = 10.0 \text{ cm } H_2O.$$

The losses represented by the admitance $Y_{wi} = R_{wi} + jL_{wi}$ are calculated separately for each section,

$$R_{wi} = S_i \cdot r_w \quad \text{and} \quad L_{wi} = S_i \cdot l_w,$$

where S_i is the circumference surface of the i-section and $r_w = 100.0$ [g/cm²s], $l_w = 0.1$ [g/cm²] — for the unit area of the circumference surface.

The slope of the voiced source frequency characteristic is assumed as −12 dB/oct. The connection of the nasal to the vocal channel is at the seventh section. All the constants mentioned can be changed in accordance with the assumed conditions of articulation.

The output signal of a given fundamental frequency F_0 was calculated by harmonic synthesis from the amplitude and phase spectra of the modelled speech sound. For its complex spectrum of the form $H(\omega) = |H(\omega)| \cdot \exp(j\phi(\omega))$, the output signal $f(t)$ is

$$f(t) = \sum_{k=1}^{N} c_k \cos(2\Pi k F_0 - \Phi_k), \tag{8}$$

where N — number of harmonics, $c_k = | U_g(\omega_k) | \cdot | H(\omega_k) |$ and $\phi_k = \arg\{(U_g(\omega_k)\} + \arg\{H(\omega_k)\}$, $U_g(\omega)$ — glottal excitation function.

3. The model of a natural vocal tract

There are many different ways of describing the vocal tract geometry. The most straightforward approach is to represent it in terms of a number of uniform cross-sectional areas specified at equidistant points from glottis to lips. Usually 17–20 areas are applied, mostly of circular shape. There were also successful attempts to apply elliptic cross-sections (e.g. FUJIMURA, 1977), but the differences between the transmission functions of these two models appeared to be insignificant. The reason of that is obvious from Eq. (2) where only the section circumfereneces S_i are different for the above two models. It is evident that augmentation of the circumference surface increases the losses related mainly to co-vibrating wall masses. This primarily has effect on q-factor of formant resonances, which is visible in modification of formant bandwiths.

When the transfer function of the vocal tract is modelled in the frequency domain, the absolute values of cross-sectional areas or their shape have a limited impact on its frequency characteristic. This is the reason why the cross-sectional areas in many models often vary in an unrealistic range, especially non an acceptable in case of continuous speech, (from 0.6 to 14 cm^2 — MRAYATI, 1976; up to 8 cm^2 — MAEDA, 1987), although the constraints imposed by the articulators on the vocal-tract shape are known (see ATAL et al., 1977). According to Atal, the cross-sectional areas vary between 0.1 and 3.5 cm^2. From our measurements and the X-ray data (BOLLA and FÖLDI, 1987) obtained for vowels articulated in the middle of the words spoken in isolation, the upper limit is a little higher (up to 4.5 cm^2).

It is obvious from several data that the vocal tract is not unform along its longitudinal axis, especially in the region of hard palate (e.g. HIKI et al., 1986). To find the geometry of the upper oral cavities limited by the hard palate and the plane tangent to the upper teeth edge, ten subjects (8 male and 2 female) has been used to make dentist plaster casts of their palates. Then, each cast was cut at every 1 cm along its midline, in planes approximately perpendicular to the mid-sagittal line of the palate. In the Fig. 4, the shapes of the cross-sections in the hard palate region determined for two subjects are presented. It is evident that the shape approximation to the circular or elliptical form is not acceptable in this case, and the trapeziform shape is more adequate. As it will be seen next, the accepted form has a significant influence on the accuracy of modelling of dynamic vocal tract cross-sections shape variations during sounds sequence articulation.

The main object of our study was to reproduce the geometry of the vocal tract of a subject uttering a given vowel and to obtain the same spectral characteristic as those for the simultaneously registered sound. As basic data we applied the X-ray pictures showing the vocal tract mid-sagittal shapes (in the side plane) registered for Polish sounds (BOLLA and FÖLDI, 1987). Besides the pictures we used the data relating to

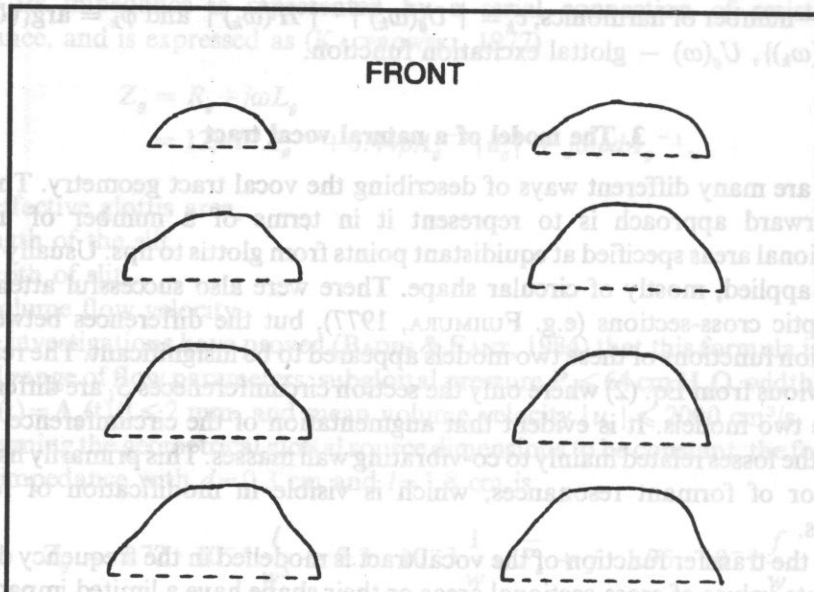

Fig. 4. Cross-sectional shapes of the oral cavity under hard palate taken for two male subjects.

distances between the referential points placed on the contours of the articulatory organs and photolabiograms with measurements.

The most difficult problem was to determine the third (lateral) dimension of the vocal tract. There were direct data only for its labial part, the cross-section shape of which was approximated by elliptical contours. The shape of the palatal cross-section of the vocal tract was assumed to be of trapeziform. In order to evaluate its dimensions, we chose (using the photo of the hard palate of the male informant participating in articulatory registration) the plaster cast of the similar dimensions and shape which fitted with reasonable accuracy (of the subject M1 from Fig. 4).

For this subject the configuration function has been determined using the measurement data concerning the distances between the reference points located on the midline of the palate and the tongue. The corresponding areas of cross-sections in palatal region are presented in the Fig. 4 (subject M1). One cm back from the rear edge of the palate a similar cross-section shape was accepted, and the shape of the cross-sections of the lower oral and pharyngeal cavities of the vocal tract, up to the glottis, was modelled as elliptical. The vertical dimension was treated as a variable depending on its location and the articulated sound, and its value was taken from the mentioned conspectus. The dimension perpendicular to the mid-sagittal plane of the vocal tract was considered as a variable independent of the articulated sound, and its variation along the longitudinal axis of this part of the vocal tract is only due to anatomy. So, for all vowels the variation of this dimension is the same. In the palatal region only the trapezium upper side size is independent of the spoken sound and remains constant. The other side, e.g. the bottom one and the height of the trapezium are variable. The main object of modelling was to reproduce for each vowel, as exactly as possible, the full mid-sagittal shape and, at the same time, its frequency characteristic evaluated from the registered signal.

4. Results of dynamic modelling

After the X-ray data, labiograms, hard palate casts and the dimensions of the pharyngeal part evaluated on the basis of formant frequencies of all vowels, the spectral characteristics were calculated for each vocal tract configuration and compared with those obtained for real speech signal. This stage of modelling was performed in order to verify these dimensions of the vocal tract which do not vary from one vowel to another.

To model the time-dependent vocal tract configuration, in case of transient sounds, we have looked for rules of control of the area function changes. We have assumed that continuous spatial changes are modelled by linear variations of all variable dimensions, in the range determined by the starting and target configurations. It is obvious that the resulting cross-shapes obtained for intermediate states of articulation are varying in the non-linear manner.

To verify our approach we modelled transient articulation between two concatenated vowels /i/ and /u/, of extreme front-back opposition. The starging and target configurations were the same as those obtained for central part of the corresponding sounds spoken in the middle of the words and applied also to model their sustained phonation version. In this example of dynamic modelling the tongue movement is extreme. The duration of the transition from the first to the second vowel was equal to eleven pitch periods (about 73 msec), the same as in natural diphthong spoken by the subject under study. For each period an intermediate configuration with its corresponding transfer function and the output signal were calculated. The pitch period time synchronization imposed by the method of harmonic synthesis enables us to observe the transient output signal period-by-period and to detect easily any non-continuous, step-like transition.

In Fig. 5 the vocal tract characteristics (amplitude and phase) calculated for two concatenated vowels /i/ and /u/ and for eleven intermediary articulatory states are presented. The transitory output signal is on the top of the next figure. It is smoothly changing from state to state, without any abrupt jumps. Similar signal calculated for a cylindrical model is presented at the bottom of the figure. The irregular variations in

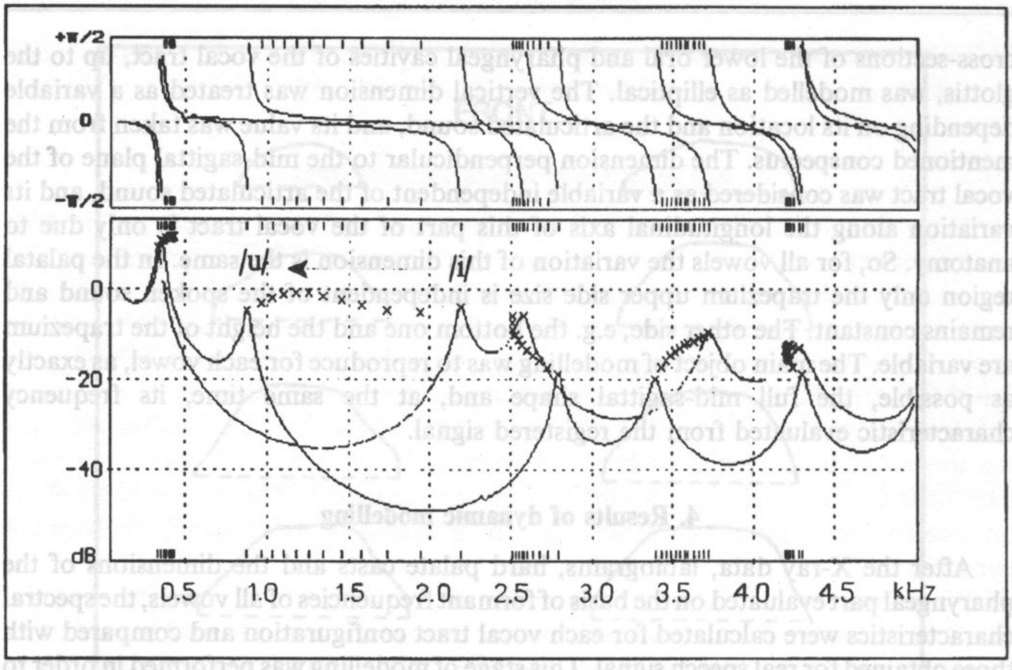

Fig. 5. Formant movements in transition from vowel /i/ to /u/ presented on phase (top) and amplitude (bottom) characteristics of vocal tract transmission function.

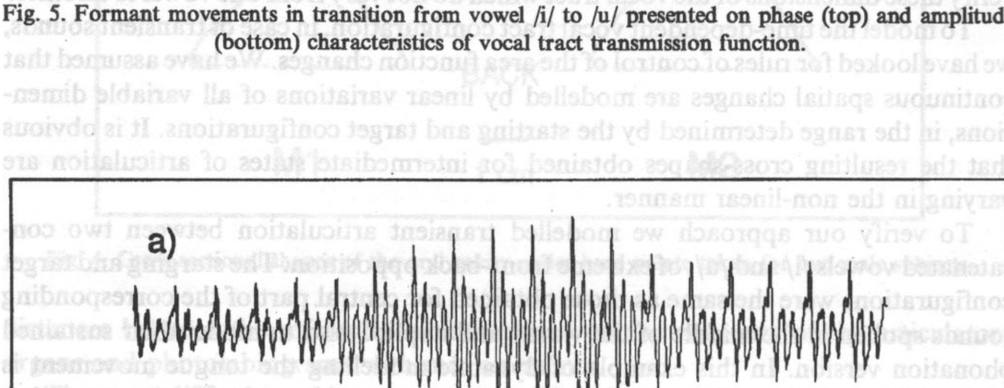

Fig. 6. Output signals for /i – u/ transition calculated for non-uniform vocal tract model (a) and circular model (b).

time are more accentuated than for the former signal. It seems that this model could not describe accurately the temporal changes in articulatory space, although the target configurations and characteristics were identical in both cases.

To compare, the formant trajectories were determined for non-uniform model and speech signal (Fig. 7). As a natural signal we analyzed a diphthong /ju/ in the Polish word "tiul" (tulle) spoken by the subject whose vocal tract was modeled. It seems that the movement of the place of articulation is rather correctly modelled. However, some

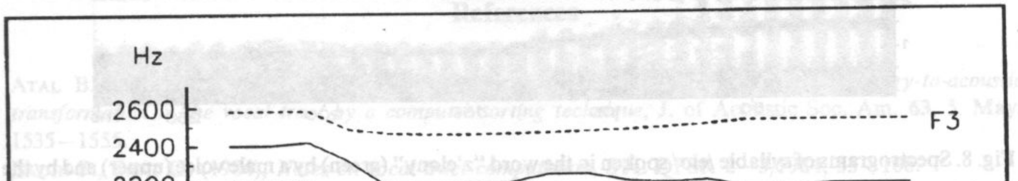

Fig. 7. Formant trajectories determined for natural (solid lines) and modelled (dashed lines) /i–u/ transition. *N* — pitch period continuous number.

discrepancies between the parameters of the modelled and the natural signal exist and this is probably due to the assumption of the movement synchronization of the whole tongue corpus. Nonetheless, for not so dramatic tongue movements the non-unform model was successful to model concatenations of vowels with different vowel-like segments (liquids, glides, for example). The example presented above could be also used to model the articulation of the syllable /ju/ with very short initial i-like segment and transition to vowel /u/.

Another example of results of dynamic modelling is presented in Fig. 8, where two spectrograms obtained for syllable /elo/ are presented. The upper spectrogram is obtained for male voice the vocal tract geometry of which was studied, the bottom picture is the result of modelling in which the vocalic target configurations (/e/, /o/ and /l/) were taken from the mentioned book (Bolla and Földi).

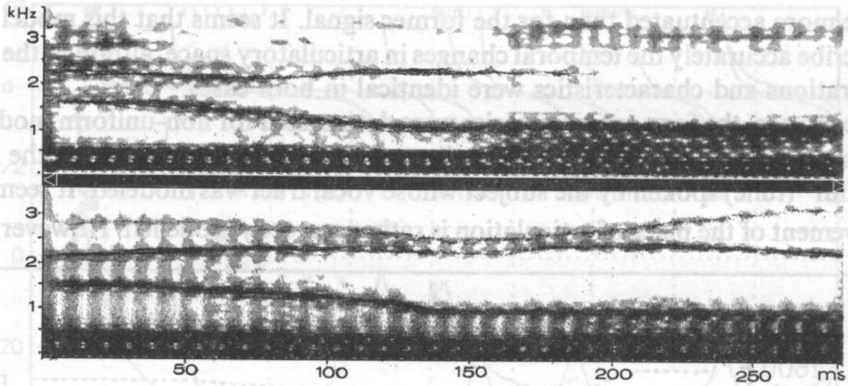

Fig. 8. Spectrograms of syllable /elo/ spoken in the word "z'elony" (green) by a male voice (upper) and by the modelled one (bottom).

The transient configurations were calculated in the same manner. It can be seen that the formant movements in the modelled syllable are very similar to those in the natural one.

5. Conclusions

We have presented an attempt to apply in speech modelling a three-dimensional vocal tract model in a more direct way. Although several simplifications were done, e.g., flat tongue surface for all vowels, trapeziform and elliptical approximations of cross-shapes, rudimentary rules of vocal tract configuration evolutions from one target to another could be applied.

The variations of lateral dimension of the vocal tract were evaluated on the basis of the plaster cast of the hard palate and labiograms. The cross-sections in the pharyngeal and laryngeal region was assumed to be identical for all vocalic sounds and have an elliptical shape with lateral dimension remaining constant for different configurations. These lateral dimensions were verified in the formant space for all six Polish vowels.

Our experiments in the development and use of three-dimensional (non-uniform) model have confirmed that it provides very effective means for the study of articulatory phenomena, especially of that connected with coarticulation pattern for different resonants. The model was also successfully applied to generate nasalized vowels and nasal consonants.

The timing characteristic of articulatory variations is linear, but it is also possible to adopt a non-linear one, e.g. of exponential form. The dynamic changes were modelled on the assumption that different articulatory dimensions vary synchronously during transitory speech waveform, from one to the following configuration. However, in the time signal a rising and sloping amplitude can be observed in the transitory part, not so accentuated in real signal. Its seems that the hypothesis of synchronous tongue movement should be verified as it was suggested by Fujimura (Fujimura, 1987), although the resulting modelled speech sounds are of very good quality.

Acknowledgment

The authors would like to express their gratitude to K. BOLLA and E. FÖLDI, researchers from the Phonetics Department at the University of Budapest, for providing us with the X-ray data and dynamic radiograms registered on videotape, without which we would not be able to carry out this research.

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PRACTICAL POSSIBILITIES OF MORE ACCURATE VELOCITY MEASUREMENTS OF ULTRASONIC WAVES IN LIQUIDS BY MEANS OF THE RESONATOR METHOD

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A precise investigation of the dependence between the resonance frequencies and the frequency for the real resonator cell was carried out using automatic measurement equipment which works in the range of frequency 0.5–10 MHz. This experimental dependence was compared with the theoretical one for the ideal resonator cell. As a result, regularities were found which are important for more accurate velocity measurements in liquids in the whole frequency range used in the resonator method. It was pointed out how on the basis of these results the method of calculating the propagation velocity in the resonator method can be improved and its accuracy increased.

1. Introduction

Physico-chemical investigations of liquid and solution properties using ultrasonic spectroscopy methods require measurements of the propagation parameters of the compression waves (attenuation and velocity) in tested liquids at a wide frequency range. Depending on the tested liquid and investigated phenomena, this range may cover the frequency from several hundreds kilocycles/s up to several gigacycles/s. As a rule it is impossible to make measurements at such a wide range using only one measurement method. The biggest obstacles occur at the beginning and at the end of the mentioned range. Wave-guide effects are an obstacle at low frequencies. At high frequencies piezoelectric transducers and electronic equipment are the main problem.

To avoid these wave-guide effects, the so-called resonator method proposed by Eggers [1–4] and, next, used and developed by other investigators [5–15] for measurements below 10 MHz has been commonly applied in the last years. This method allows to measure the attenuation of a tested liquid as well as the propagation velocity of an ultrasonic wave as a function of the frequency. The attenuation coefficient is determined by the measurement of the quality factor of the resonator cell with the tested liquid inside. The ultrasonic velocity is calculated from the frequency distance between the subsequent resonance frequencies of the resonator cell (filled up with the tested liquid). A theory about this method can be found in the literature [1, 2, 6, 9–11].

The measurements of attenuation do not cause particular difficulties, but those of velocity are more controversial.

From the theoretical point of view, the principle of velocity measurement is very simple. In accordance with [1, 2], in the ideal resonator cell the frequency intervals between subsequent resonances Δf_n, should be the same and equal to the first basic resonance frequency of that system, f_1:

$$\Delta f_n = f_{n+1} - f_n = f_1 = \frac{c}{2l}, \quad n=1,2,... \tag{1}$$

where c is the wave velocity in the liquid, l is the distance between the internal surfaces of the transducers, n is the integer, a number of the resonance.

Thus when the distance l is known and the interval Δf_n is measured, one can find velocity c from Eq. (1).

In practice, the relationship between Δf_n and f_1 is complicated and direct use of Eq. (1) to determine the velocity causes significant errors. Indeed, many investigators [1, 2, 6, 10, 11] noticed that resonance frequencies are not to be situated in the equal frequency distances in the real made resonator cell. Especially at frequencies close to f_q (where f_q is the fundamental resonance frequency of the transducers) and odd harmonic, e.g. $3 f_q$, $5 f_q$ etc. Δf_n are smaller than f_1 in a significant manner.

EGGERS [1, 2] presented the formula for calculating the direct value of velocity from measurements of the frequency intervals between subsequent resonances (Δf_n) but that may be used only in the frequency range that is close to the anti-resonance frequencies of the transducers of the resonator cell, e.g. $\frac{1}{2} f_q$, $\frac{3}{2} f_q$ etc. without a definition of the applicability range.

SARVAZAN [9] also considered the velocity measurements but limited his considerations only to relative measurements. LABHARDT [6] obtained an analytical relationship for the direct calculation of velocity from the measurements of Δf_n; however, it requires some of the values of the resonator cell parameters, the determination of which, mostly in the experimental way, introduce additional errors.

Thus, as one can notice, in the literature no precise investigations of the dependence between Δf_n and the frequency f for the real resonator cells can be found. Such investigations are fundamental for accurate velocity measurements and, what is very important, they enable to employ the whole frequency range used in the resonator method not only at the anti-resonance frequencies like in [1, 21].

The aim of this work was to study the dependence between Δf_n and the frequency f for the resonance cells and as a result of this investigation, determine the consequences for the velocity measurements and their accuracy.

2. Formulation of the problem

The solutions mentioned by EGGERS [2] were the starting point for the study. According to those, the first resonance frequency f_1 for the liquid layer in the resonator cell can be determined with great accuracy by measuring f_n and Δf_n at a frequency close to $\frac{f_q}{2}$ (where f_q is the fundamental resonance frequency of the transducers used in the resonance cell) and, subsequently, by introducing the correction which accounts for the influence of the transducers on the measured frequency of the resonance. Under these conditions

$$f_1 = \frac{f_n}{n}\left[1 + \frac{R}{n}\left(\frac{2f_n}{f_q} - 1\right)\right], \tag{2}$$

where R is the ratio of acoustical impedances of the tested liquid and the quartz (the material of the transducers), n is the number of resonance.

For the frequency f close to $\frac{f_q}{2}$, n is calculated from Eq. (3)

$$n \cong \frac{f_n}{\Delta f_n} \tag{3}$$

making the result even to an integral.

After calculating f_1, one can find the velocity from the relationship (4)

$$c = 2\,l f_1. \tag{4}$$

However, in [2] there was no information as to what range of frequency the correction introduced in Eq. (2) allows to get the value of f_1 and, consequently, to get the value of c, with order of the accuracy of about 0.1%. Such an exactness is regarded to be the minimum accuracy required in physico-chemical investigations.

In the frequency range distant to the anti-resonance frequency of the tranducers, the values of $\frac{f_n}{n}$ are different in a significant manner from f_1 and the correction introduced in Eq. (2) is not sufficient. On the other hand, when n is determined from the relationship (3) a significant error is also involved.

The experimental investigations carried out and described below allow to determine the range of applicability of the calculation procedure based on Eqs. (2) and (3). These measurements enable to calculate f_1 in a different, more accurate way in the whole measurement frequency range.

3. Measuring system

When the method described above is used in measurements, the transducer of the resonator must have electric signals from frequency source that can supply stable frequency and have the possibility of frequency change in the range defined by the resonator. The output signal, sometimes on the millivolts level, is very sensitive to disturbances. In the article [14] the authors describe a set-up for ultrasonic measurements using the resonator method with a manually operated frequency synthesizer as the supply source of the resonator cell. The output signal was amplified in the selective amplifier especially designed for this purpose and was indicated on a typical oscilloscope. However, measurements in this set-up were time-consuming, the amplification of the receiving channel was not sufficient for frequency distant from the resonance frequency of the resonator transducers and the determination of the signal level of the output signal from the oscilloscope was not so accurate.

To eliminate these inconveniences, a new equipment was constructed [15], which is shown in the block diagram in Fig. 1. This equipment automatically changes the frequency of the supplying signal. Disturbances of the received signal are eliminated by using a superheterodyne circuit in the receiving channel. The electronic part of the equipment may work in the frequency range 20 kHz – 10 MHz. However, the measurement possibilities of the resonators limit the lower range of the frequencies to 0.5 MHz.

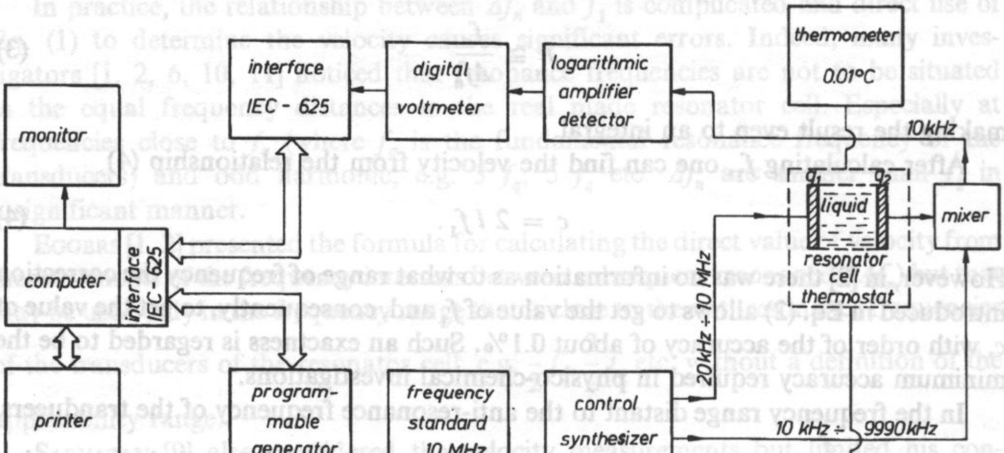

Fig. 1. The block diagram of the overall set-up.

The Hewlett-Packard mod. 3324A generator with a minimal step 1 mHz and range from 1 mHz to 60 MHz was used as the programmable source of the variable frequency. It supplied the synthesizer which controlled the resonator and delivered voltage to the mixer. The synthesizer, the mixer and the amplifier were made to be used in this measuring equipment. The digital voltmeter Meratronik mod. V553 was used as A/D converter from which data were transmitted to an IBM PC/AT computer. It gave 0.01 dB resolution in this equipment.

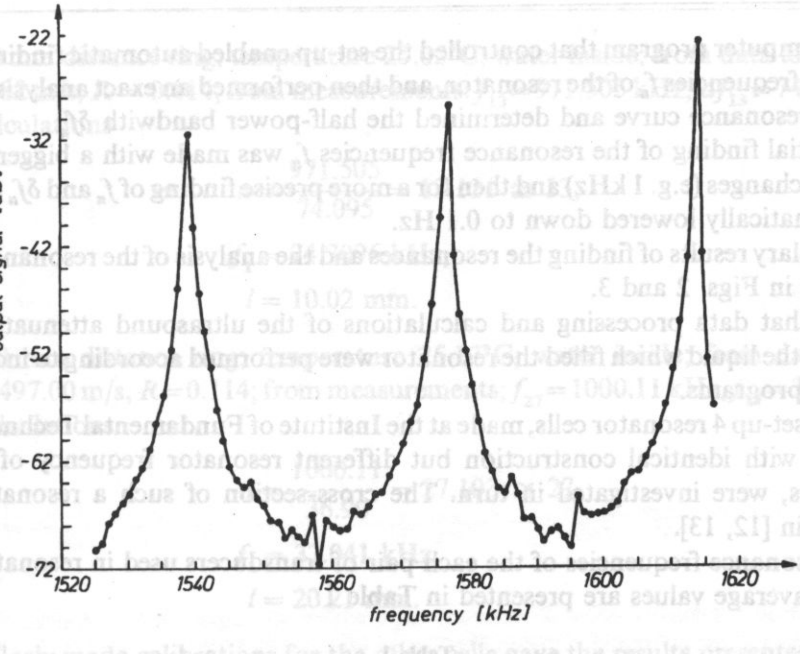

Fig. 2. The resonance peaks of the resonator.

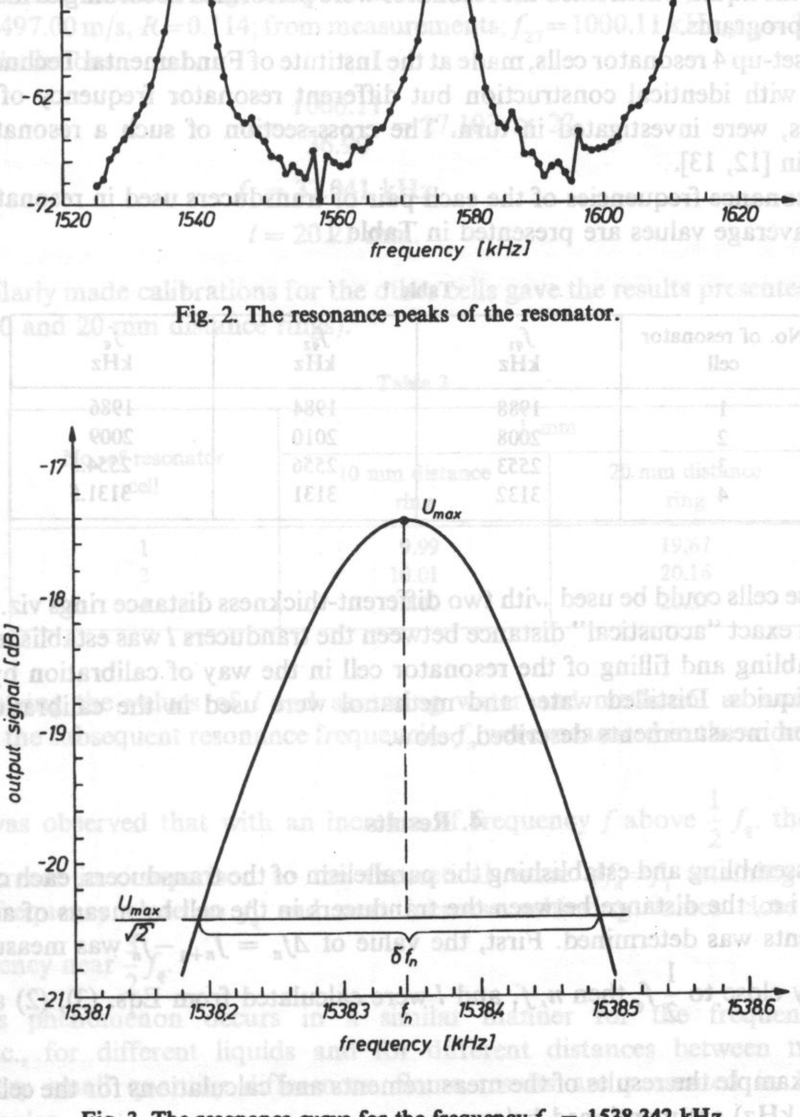

Fig. 3. The resonance curve for the frequency f_n = 1538,342 kHz.

The computer program that controlled the set-up enabled automatic finding of the resonance frequencies f_n of the resonator, and then performed an exact analysis of each obtained resonance curve and determined the half-power bandwith δf_n.

An initial finding of the resonance frequencies f_n was made with a bigger step of frequency changes (e.g. 1 kHz) and then for a more precise finding of f_n and δf_n this step was automatically lowered down to 0.1 Hz.

Exemplary results of finding the resonances and the analysis of the resonance curve are shown in Figs. 2 and 3.

After that data processing and calculations of the ultrasound attenuation and velocity in the liquid which filled the resonator were performed according to individual computer programs.

In this set-up 4 resonator cells, made at the Institute of Fundamental Technological Research, with identical construction but different resonator frequency of quartz transducers, were investigated in turn. The cross-section of such a resonator was presented in [12, 13].

The resonance frequencies of the each pair of transducers used in resonator cells and their average values are presented in Table 1.

Table 1

No. of resonator cell	f_{q1} kHz	f_{q2} kHz	f_q kHz
1	1988	1984	1986
2	2008	2010	2009
3	2553	2556	2554.5
4	3132	3131	3131.5

All these cells could be used with two different-thickness distance rings viz. 10 and 20 mm. An exact "acoustical" distance between the tranducers l was established after each assembling and filling of the resonator cell in the way of calibration by using reference liquids. Distilled water and methanol were used in the calibration and investigation measurements described below.

4. Results

After assembling and establishing the parallelism of the transducers, each cell was calibrated, i.e., the distance between the tranducers in the cell by means of acoustic measurements was determined. First, the value of $\Delta f_n = f_{n+1} - f_n$ was measured at a frequency close to $\frac{1}{2} f_q$ then n, f_1 and l were calculated from Eqs. (3), (2) and (4) respectively.

As an example the results of the measurements and calculations for the cell No. 2 ($f_q = 2009$ kHz) are mentioned below:

1) 10 mm distance ring; temperature 25.05°C; water inside; from data tables: c_{water} = 1496.82 m/s, R = 0.114; from measurements: f_{13}=971.505 kHz, Δf_{13}=74.095 kHz; from calculations

$$n = \frac{971.505}{74.095} = 13.111 \cong 13,$$

$$f_1 = 74.7096 \text{ kHz},$$

$$l = 10.02 \text{ mm}.$$

2) 20 mm distance ring; temperature 25.17°C; water inside; from data tables: c_{water}=1497.00 m/s, R=0.114; from measurements; f_{27}=1000.11 kHz, f_{27}=36.90 kHz; from calculations

$$n = \frac{1000.11}{36.90} = 27.103 \cong 27,$$

$$f_1 = 37.041 \text{ kHz},$$

$$l = 20.21 \text{ mm}.$$

Similarly made calibrations for the other cells gave the results presented in Table 2 (for 10 and 20 mm distance rings).

Table 2

No. of resonator cell	l, mm	
	10 mm distance ring	20 mm distance ring
1	9.99	19.67
3	10.01	20.16
4	10.00	20.27

Knowing the values of l and assuming water and methanol as nondispersive liquids, the subsequent resonance frequencies f_n were measured in the wide frequency range.

It was observed that with an increase of frequency f above $\frac{1}{2} f_q$, the intervals Δf_n decrease in comparison to the theoretical value $\Delta f_n = f_1$ attaining minimum for a frequency close to f_q and next increase achieving values close to f_1 for a frequency near $\frac{3}{2} f_q$.

This phenomenon occurs in a similar manner for the frequencies 3 f_q, 5 f_q etc., for different liquids and for different distances between transducers with very small quantity differences. Some results are presented in Figs. 4–7 as examples.

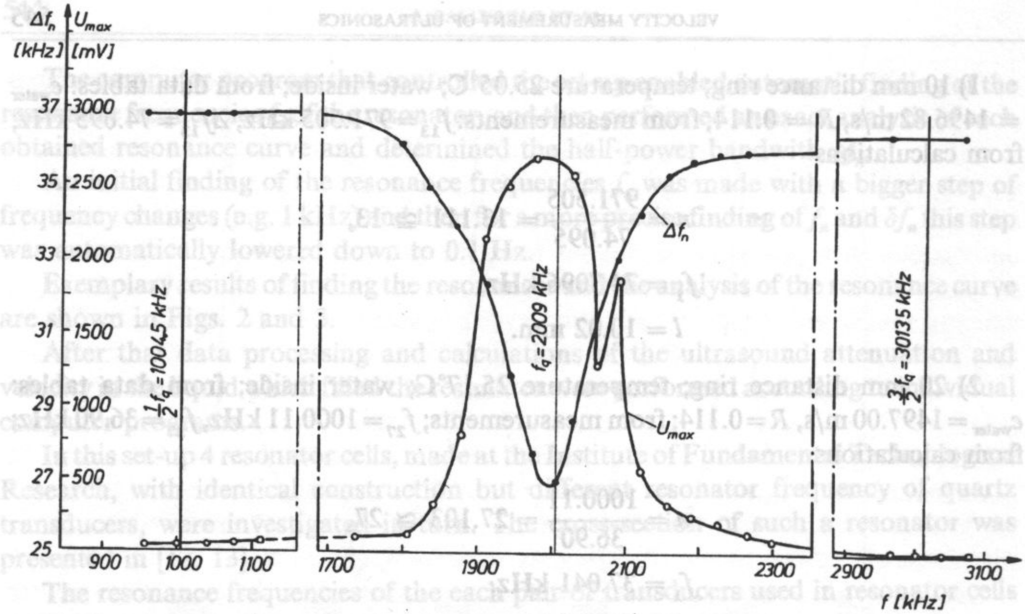

Fig. 4. The frequency interval Δf_n and maximum amplitude of the output signal at resonances U_{max} as a function of the frequency f for the cell No. 2. The 20 mm distance ring. Water. $f_q = 2009$ kHz, $l = 20.21$ mm, $f_1 = 37.041$ kHz, $U_{input} = 5.8$ V.

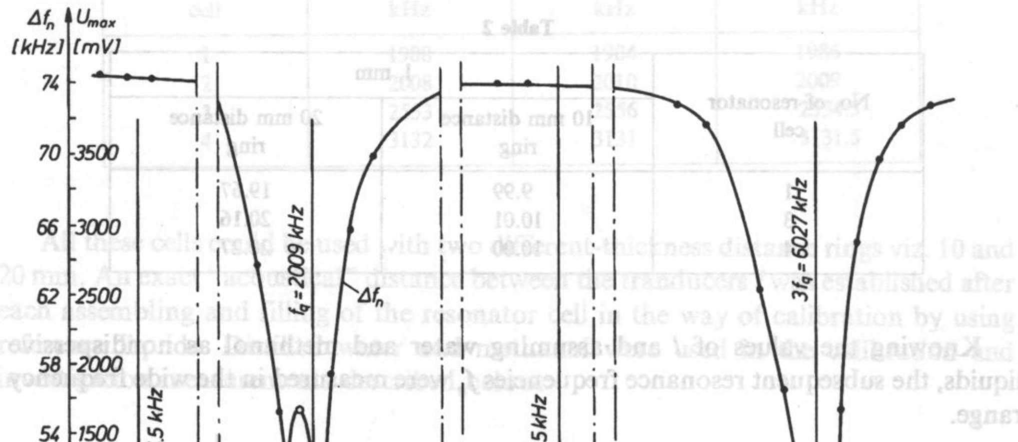

Fig. 5. The frequency intervals Δf_n and maximum amplitude of the output signal at resonances U_{max} as a function of the frequency f for the cell No. 2. The 10 mm distance ring. Water. $f_q = 2009$ kHz, $l = 10.02$ mm, $f_1 = 74.7096$ kHz, $U_{input} = 5.8$ V.

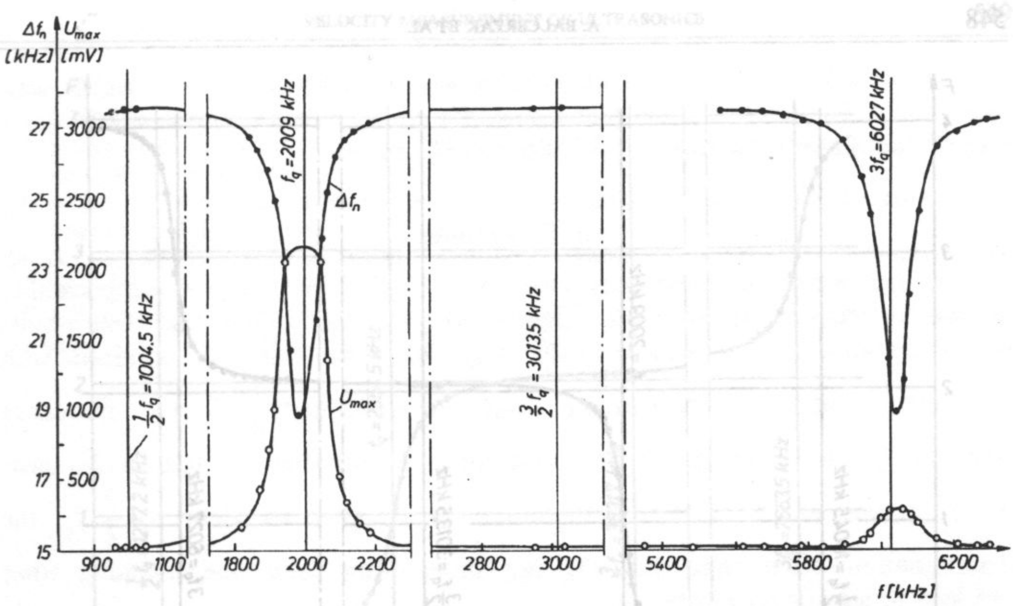

Fig. 6. The frequency intervals Δf_n and maximum amplitude of the output signal at resonances U_{max} as a function of the frequency f for the cell No. 2. The 20 mm distance ring. Methanol. $f_q = 2009$ kHz, $l = 20.06$ mm, $f_1 = 27.571$ kHz, $U_{input} = 5.8$ V.

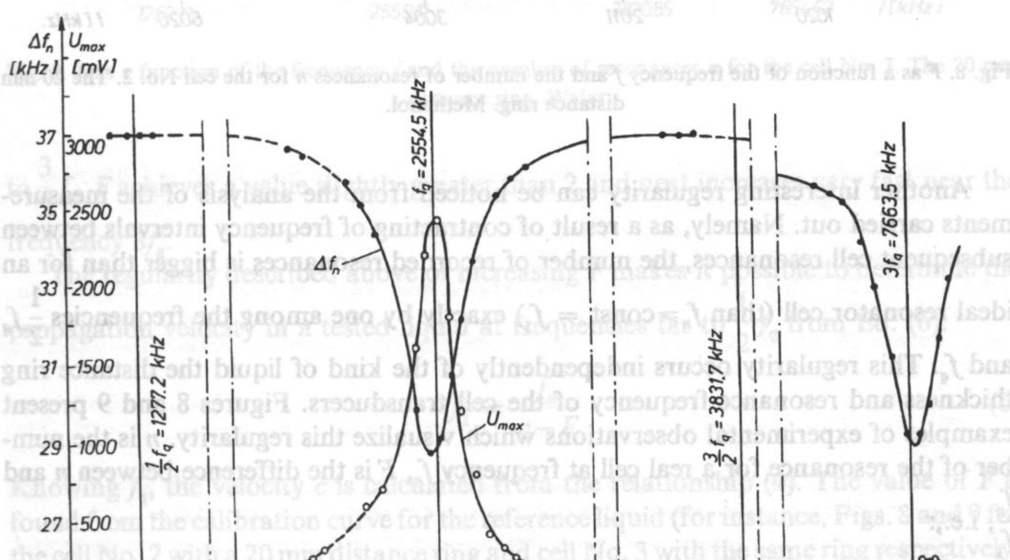

Fig. 7. The frequency intervals Δf_n and maximum amplitude of the output signal at resonances U_{max} as a function of the frequency f for the cell No. 3. The 20 mm distance ring. Water. $f_q = 2554.5$ kHz, $l = 20.16$ mm, $f_1 = 37.121$ kHz, $U_{input} = 5.8$ V.

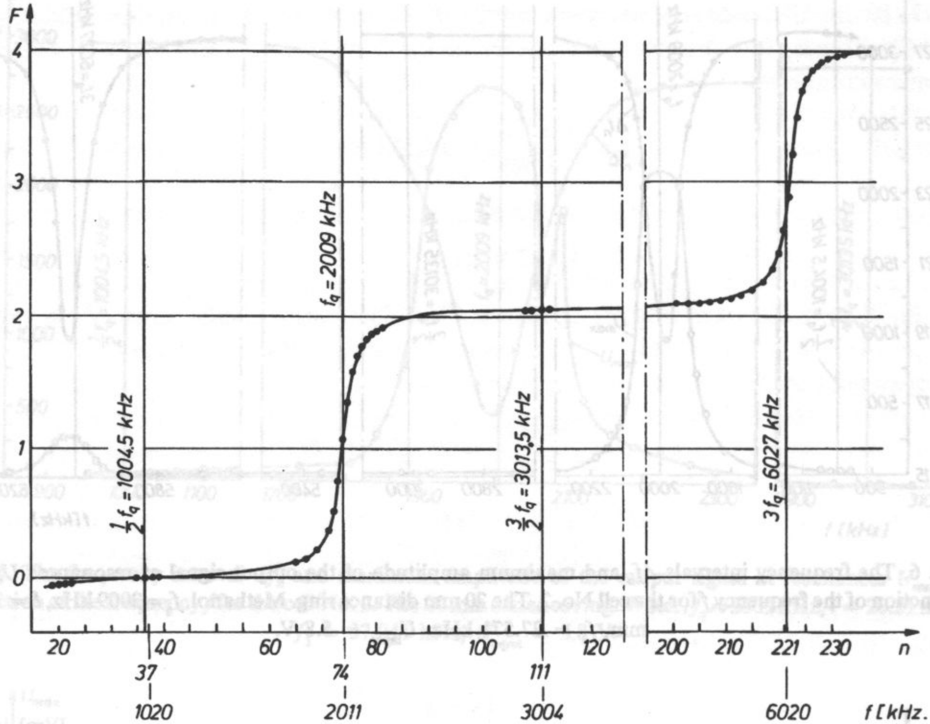

Fig. 8. *F* as a function of the frequency *f* and the number of resonances *n* for the cell No. 2. The 20 mm distance ring. Methanol.

Another interesting regularity can be noticed from the analysis of the measurements carried out. Namely, as a result of contracting of frequency intervals between subsequent cell resonances, the number of recorded resonances is bigger than for an ideal resonator cell (than $f_n = \text{const} = f_1$) exactly by one among the frequencies $\frac{1}{2} f_q$ and f_q. This regularity occurs independently of the kind of liquid the distance ring thickness and resonance frequency of the cell transducers. Figures 8 and 9 present examples of experimental observations which visualize this regularity. n is the number of the resonance for a real cell at frequency f_n, F is the difference between n and $\frac{f_n}{f_1}$, i.e.,:

$$F = n - \frac{f_n}{f_1}. \tag{5}$$

The ratio $\frac{f_n}{f_1}$ is the number of resonances for an ideal resonator cell; this is a real number. Thus, as it was mentioned above, the value of F equals 1 at $f = f_q$. At frequencies close

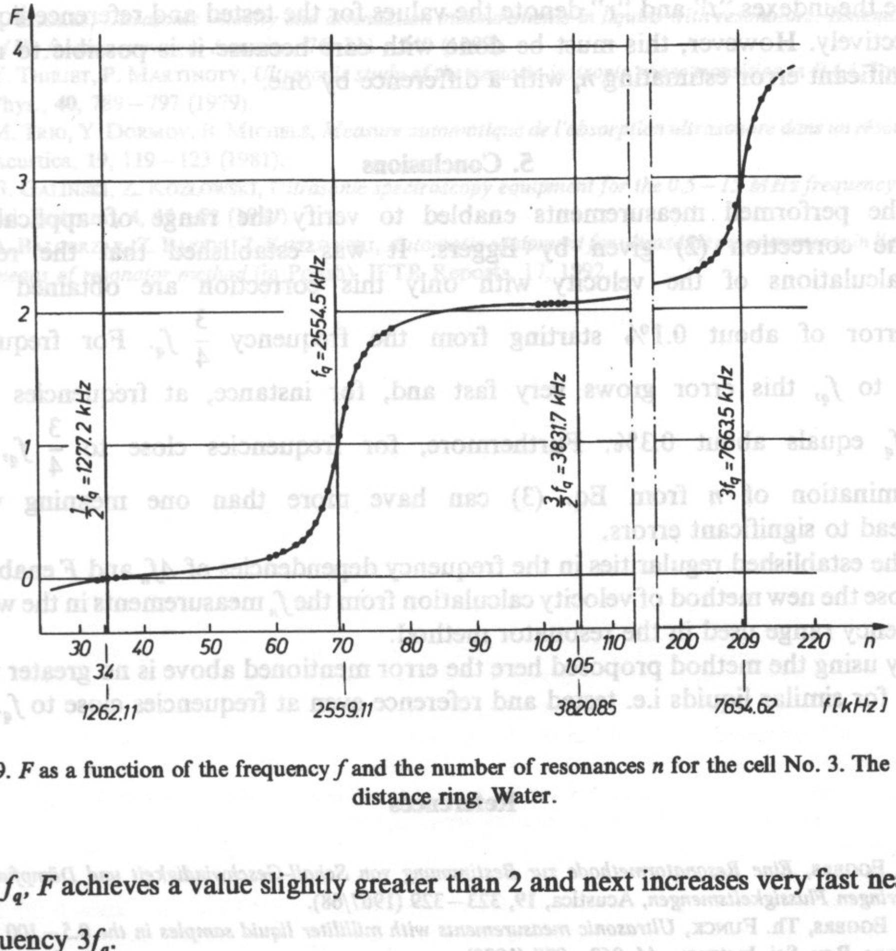

Fig. 9. F as a function of the frequency f and the number of resonances n for the cell No. 3. The 20 mm distance ring. Water.

to $\frac{3}{2}f_q$, F achieves a value slightly greater than 2 and next increases very fast near the frequency $3f_q$.

The regularity described above of increasing F makes it possible to determine the propagation velocity in a tested liquid at frequencies far to $\frac{1}{2}f_q$ from Eq. (6).

$$f_1 = \frac{f_n}{n-F} \tag{6}$$

Knowing f_1, the velocity c is calculated from the relationship (4). The value of F is found from the calibration curve for the reference liquid (for instance, Figs. 8 and 9 for the cell No. 2 with a 20 mm distance ring and cell No. 3 with the same ring respectively) at the frequency of that measurement. One can encounter difficulties when determining the number of real resonance for the tested liquid n_t, which should be estimated from Eq. (7)

$$n_t \cong \frac{(f_n)_t}{(f_1)_r} \tag{7}$$

where the indexes "t" and "r" denote the values for the tested and reference liquids, respectively. However, this must be done with care because it is possible to make a significant error estimating n_t with a difference by one.

5. Conclusions

The performed measurements enabled to verify the range of applicability of the correction (2) given by Eggers. It was established that the results of calculations of the velocity with only this correction are obtained with an error of about 0.1% starting from the frequency $\frac{3}{4} f_q$. For frequency close to f_q, this error grows very fast and, for instance, at frequencies near 0.9 f_q equals about 0.3%. Furthermore, for frequencies close to $\frac{3}{4} f_q$, the determination of n from Eq. (3) can have more than one meaning what can lead to significant errors.

The established regularities in the frequency dependencies of Δf_n and F enable to propose the new method of velocity calculation from the f_n measurements in the whole frequency range used in the resonator method.

By using the method proposed here the error mentioned above is no greater than 0.1% for similar liquids i.e. tested and reference even at frequencies close to f_q.

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BIAS AND STANDARD DEVIATION OF DIGITAL MEAN AND MAXIMUM DOPPLER FREQUENCY ESTIMATORS

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The performance of four spectral techniques (FFT, AR Burg, ARMA and Arithmetic Fourier Transform AFT) for mean and maximum frequency estimation of the Doppler spectra is described. The mean frequency was computed as the first spectral moment of the spectrum with and without the noise subtraction. Different definitions of f_{max} were used: frequency at which spectral power decreases down to 0.1 of its maximum value, modified threshold crossing method [23] and novel geometrical method. "Goodness" and efficiency of estimators were determined calculating the bias and standard deviation of the estimated mean and maximum frequency of the computer simulated Doppler spectra. The power of analysed signals was assumed to have the exponential distribution function. The SNR ratios were changed over the range from 0 to 20 dB. The AR and ARMA models orders selections were done independently according to Akaike Information Criterion (AIC) and Singular Value Decomposition (SVD). It was found, that the ARMA model computed according to SVD criterion had the best overall performance and produced the results with the smallest bias and standard deviation. The preliminary studies of the AFT proved its attractiveness in real-time computation, but its statistical properties were worse than that of the other estimators. It was noticed that with noise subtraction the bias of f_{mean} decreased for all tested methods. The geometrical method of f_{max} estimation was found to be more accurate of other tested methods, especially for narrow band signals.

1. Introduction

Doppler ultrasound is widely used technique for measuring of blood flow in vessels. The proper estimation of mean and maximum Doppler frequency is crucial for flow quantification. The mean Doppler frequency (f_{mean}) carries the information on the mean blood flow velocity; for known vessel diameter, the volumetric flow can be then computed. The detection of the maximum frequency (f_{max}) is a good indicator of narrowing of the vessel. The tighter the stenosis is, the higher is the expected velocity.

Different methods of the estimation of the Doppler frequencies in the time and frequency domains are described in literature [4, 17, 25, 28]. The performance of both parametric and non-parametric spectral estimators was done by VAITKUS et al. [38], but there is however, a lack of analysis of assessment of the influence of the applied spectral estimation methods on maximum Doppler frequency measurement.

The influence of four methods used for modelling of the power spectral density (PSD) on mean and maximum frequency estimation is described in this paper. The examined methods were: FFT, AR Burg, ARMA and arithmetic Fourier transform (AFT). For all mentioned spectrum estimators the mean frequency was computed as the first moment of the spectrum with and without noise subtraction. The novel geometrical method of f_{max} determination is described and compared with other methods described by d'Alessio [12] and Mo et al. [23]. The evaluation was done using computer simulated stationary Doppler signals with added white noise.

Performance of examined estimators was determined by calculating bias and standard deviation of f_{mean} and f_{max}

2. Doppler signal modelling

The early theoretical and experimental work of Shung et al. [30] showed that the scattering of ultrasound in blood was proportional to the fourth power of frequency.

The scattering was found to depend upon the hematocrit HTC of blood. The increase of scattering was observed for HTC up to 24% reaching the plateau between 24% and 30% and next decrease occured. Shung et al. [30] concluded that for HTC greater than 20% the assumption of the independent backscattering was rather unrealistic. Their results were in good agreement with Twersky's [36] wave scattering theory for small scatteres and heuristic "hole" approach [30].

A number of works on multiple scattering of acoustic waves in media with a small fractional volume were published [15]. The higher order approximation [21] for higher fractional volume of scatterers resulted in better understanding of scattering phenomena. Another approach was presented by Angelsen [3] assuming that the backscattered signal originated from the variations of the local density and compressibility of blood.

However, the existing theoretical descriptions are not complete to incorporate the effect of multiple scattering for the ultrasonic field parameters. For this reason we have neglected in this work multiple scattering and the effect of the interaction between the red cells in blood. The applied model is similar to the one used by Atkinson and Woodcock [5], Ferrara [14] and Holland [16]. We have neglected such factors as beam sensitivity and attenuation in the tissue.

Let the transmitted signal be of the form

$$s(t) = A\cos(\omega_0 t), \tag{1}$$

where ω_0 is the radial transmitted frequency.

The ultrasonic signal backscattered from the population of the red cells can be approximated by

$$f(t) = \sum_i A_i \cos(\omega_i t + \varphi_i), \tag{2}$$

where A_i and φ_i are respectively amplitudes and phases of the echo backscattered at i-th scatterer.

Since the backscattered Doppler echo is a single-sideband like signal (with suppressed carrier) it is convenient to represent it by analytical signal $Z(t)$ derived from the real function $f(t)$ using the Hilbert transform $F_H(t)$.

Function $F_H(t)$ is a quadrature function of $f(t)$ because all of their frequency components are shifted by $\Pi/2$ with respect to $f(t)$.

In polar coordinates $Z(t)$ can be written as

$$Z(t) = |Z(t)|\, e^{\omega_i t + \varphi_i},$$

where

$$|Z(t)| = \sqrt{f^2(t) + F_H^2(t)}\,,$$
$$\omega_i t + \varphi_i = \arctan\left(\frac{F_H(t)}{f(t)}\right). \tag{3}$$

The amplitude of analytical signal determines its instantaneous amplitude or envelope while time rate of change of its phase is the instantaneous frequency.

Quadrature components $f(t)$ and $F_H(t)$ are the real and imaginary parts of the input signal. The frequency of input signal is increased or decreased by a value of Doppler shift ω_d originated at moving particles.

After mixing $Z(t)$ with reference signals $\sin(\omega_0 t)$ in one channel and $\cos(\omega_0 t)$ in second channel two base band quadrature signals are obtained

$$I(t) = f(t)\cos(\omega_0 t) + jF_H(t)\sin(\omega_0 t)$$
$$Q(t) = -f(t)\sin(\omega_0 t) + jF_H(t)\cos(\omega_0 t) \tag{4}$$

Since the position of blood particles is random, $I(t)$ and $Q(t)$ components of the backscattered signal are random variables. Each particle is an independent source of echo with random phase φ_i, Doppler phase $\omega_i t$ and amplitude A_i.

For stationary targets the random phase is equal to

$$\varphi_i = \frac{4\pi d_i}{\lambda}, \tag{5}$$

where d_i is the distance from the transducer to the i-th scatterer and λ is the wavelength, $\lambda = c/f_0$, c is the speed of sound in blood.

When particles are moving with radial velocity v_{ri} towards (or away) the transducer then the rate of change of phase, called the Doppler shift, is equal to

$$\frac{d\varphi_i}{dt} = \frac{4\pi v_{ri}}{\lambda} = \omega_{di}. \tag{6}$$

Both stationary random phase and Doppler phase can be combined together. After replacing φ_i in Eq. (2) by $\varphi_i + \omega_{di} t$. the demodulated analytical signal $Z_d(t)$ becomes

$$Z_d(\mathrm{t}) = \sum_i A_i \cos(\omega_{di} t + \varphi_i) + j\sum_i A_i \sin(\omega_{di} t + \varphi_i). \tag{7}$$

The insonified field or sample volume is much larger than the wavelength (ct >> λ). For large number of scatterers in the sample volume, the probability distribution of φ_i is expected to have uniform density function within the range $-\Pi, +\Pi$. In fact, between the minimum and maximum values of d_i, φ_i spans over many intervals of 2Π, folding into the $-\Pi$ and $+\Pi$ range, justifying even better the assumption of uniform distribution.

The central limit theorem states that the distribution of sufficiently large sum of independent random variables approaches a Gaussian distribution.

Applying this theorem to the Eq. (7) we can conclude that both real $I(t)$ and imaginary $Q(t)$ components of complex envelope $Z(t)$ are independent random variables and have Gaussian distribution with zero mean. It yields that the distribution of amplitude A of complex envelope is equal to the joint probability distribution of $I(t)$ and Q(t)

$$p[I(t),Q(t)] = p[I(t)]\,p[Q(t)] = \frac{1}{2\pi\sigma^2}\, e^{-\frac{(I^2(t)+Q^2(t))}{2\sigma^2}}, \tag{8}$$

Replacing the variables $I(t)$ and $Q(t)$ by their equivalents in the polar coordinates, $A = \sqrt{(I^2+Q^2)}$ and $\varphi = \arctan(I/A)$, the joint density function becomes

$$p[A,\varphi] = \frac{A}{2\pi\sigma}\, e^{-\frac{A^2}{2\sigma^2}}, \tag{9}$$

The distribution of φ_i alone is uniform and equal to $1/(2\Pi)$ over $\langle -\Pi..\Pi \rangle$.

The distribution of A alone is given by

$$p[A] = \frac{A}{\sigma^2}\, e^{-\frac{A^2}{2\sigma^2}} \tag{10}$$

for $A > 0$.

The simulated Doppler signal should have the statistical properties identical or at least similar to that of the backscattered ultrasonic signal from the blood. The time averaged spectral density of C.W. Doppler (continuous wave Doppler) signal for parabolic blood velocity profile is uniform within the range from $f_{\min}$ up to $f_{\max}$ corresponding to the minimum and maximum velocity of blood [7]. For pulse Doppler, the backscattered energy returns from a small sample volume and the time averaged spectrum is considerably narrowed. Also for C.W. Doppler, combination of narrow ultrasonic beam and flat blood velocity profile result in narrow Doppler spectrum. Without introducing a substantial error the envelopes of those spectra can be approximated by the Gaussian functions. The similar approach was used in modelling of radar precipitation signals [13, 31].

The resulting spectral envelope becomes

$$G_n = \frac{1}{\sigma\sqrt{2\pi}}\, e^{-\frac{(f_n-\mu)^2}{2\sigma^2}}, \tag{11}$$

where G_n is the discrete spectral coefficient at frequency f_n, μ is a power weighted mean frequency and σ is the standard deviation of the spectrum.

This approach facilitates control over the simulated mean frequency and the bandwith 2σ of the Doppler spectrum.

The noise present in the Doppler signal is assumed to be a white one. The amplitude of the backscattered ultrasound has Rayleigh distribution (see Eq. (10)). It was shown [31, 32] that the amplitude of sum of the signal and noise must have Rayleigh distribution function and its power P_n must be exponentialy distributed

$$p[P_n] = \frac{1}{\sigma_s^2} e^{-\frac{P_n}{\sigma_s^2}}, \tag{12}$$

where σ_s^2 equals to the average power of the signal.

The generation of random variable having a particular distribution is accomplished using an inverse cumulative distribution function (CDF) transformation [22, 26]. If a source of uniformly distributed random variables x_n is presupposed then a random variable P_n are obtained according to

$$P_n = \sigma_s^2 \ln(x_n). \tag{13}$$

The proper scaling of signal and noise depends on SNR value defined as

$$\mathrm{SNR} = 10 \log_{10} \left(\frac{\sum_n G_n}{\sigma_{ns}^2} \right), \tag{14}$$

where ΣG_n is the power of the signal and σ_{ns}^2 is noise power.

Now, we may replace average power σ_s^2 in (13) by an expression including the spectral envelope G_n and SNR [31].

$$\sigma_s^2 = \frac{\sigma_{ns}^2 10^{\frac{\mathrm{SNR}}{10}}}{\sum_n G_n} G_n + \frac{\sigma_{ns}^2}{N} \tag{15}$$

and finally we arrive at the expression describing the composite spectral density outlined by Gaussian envelope G_n

$$P_n = \left(\frac{\sigma_{ns}^2 10^{\frac{\mathrm{SNR}}{10}}}{\sum_n G_n} G_n + \frac{\sigma_{ns}^2}{N} \right) \ln(x_n), \tag{16}$$

The next step is to obtain the real A_n and imaginary B_n components of the composite spectrum. The amplitudes of real part A_n and imaginary part B_n of the composite spectrum are Gaussian distributed

$$A_n = \sqrt{P_n}\cos(2\pi x_n)$$
$$B_n = \sqrt{P_n}\sin(2\pi x_n). \tag{17}$$

The properties of CDF transformation were applied here again; the products of Rayleigh distributed $\sqrt{P_n}$ and cos $(2\Pi x_n)$ or sin $(2\Pi x_n)$ where x_n is uniformly distributed over the range {0,1}, generate two independent Gaussian distributed processes.

Finally, the quadrature time signals are obtained by inverse Fourier transform of the composite spectrum

$$\mathscr{F}^{1}\{A_n+jB_n\} = I(t)+jQ(t). \tag{18}$$

The variable parameters of simulated Doppler spectra were: mean frequency, maximum frequency, bandwith and signal to noise ratio SNR, varied from 0 to 20 dB (0, 3, 6, 10, 20 dB). The sampling rate was set to 20 kHz. The signal was generated according to (17) using a 1024 points FFT, and only the real part of the signal was considered in further analysis. Every realization of such random processes was simulated by a block of 3072 adjacent samples. Those time series were observed using a 128 pts. Hamming window in order to achieve local stationarity of the signal and to reduce Gibbs phenomena.

The mean frequency of the Gaussian enveloped spectrum was changed from 1250 Hz to 7000 Hz. Different spectra widths were controlled by setting standard deviation σ (Eq. (14)) equal to 16, 32 and 64 points or, in frequency units, 320 Hz, 640 Hz and 1280 Hz.

Along with the Gaussian enveloped spectra the rectangular spectra of 320 Hz and 1280 Hz bandwidth were generated in order to simulate the parabolic flow insonified uniformly by C.W. Doppler.

3. Estimation of PSD

3.1. Periodogram

The periodogram was chosen as the reference method. The power spectral density function (PSD) was estimated as:

$$P_{\mathrm{PER}}(f) = \frac{1}{N}\left|\sum_{n=0}^{N-1} x(n)\, e^{-j2\Pi fn}\right|^2. \tag{19}$$

In spite of well known advantages of the periodogram (known statistics, medium computational cost, good performance for noisy signals), this PSD estimator performance suffers from both an increased variance (regardless of the observation time window), as well as spectral leakage due to the implicit windowing of the data. In analysis of the Doppler signals it causes an unreliable f_{mean} and $f_{\max}$ estimation for low SNRs. Reduction of the estimator's variance, through averaging of succeeding or overlapping data segments is one of the remedy, however limited for real-time signal processing.

3.2. Periodogram via Arithmetic Fourier Transform (AFT)

REED et al. [27] proposed an algorithm of Fourier coefficients determination with linear computational cost (only one multiplication per Fourier coefficient with number of addition comparable to FFT). This very efficient method is based on inverse Mobius theorem. According to this theorem, every function $f(n)$, not vanishing in $<1, N>$ and vanishing outside $<1,N>$ ($f(n) = 0$, for $n>N$) may be expressed as

$$f(n) = \sum_{m=1}^{\mathrm{trunc}(N/n)} \mu(m)\, g(mn),$$

where (20)

$$g(n) = \sum_{k=1}^{\mathrm{trunc}(N/n)} f(kn).$$

with $\mu(m)$ denoting Mobius function and trunc(x) denoting integer part of real number x.

Then the real continuous signal $A(t)$ with zero mean, has n-th average shifted by α defined as

$$S(n,\alpha) = \frac{1}{n}\sum_{m=0}^{N-1} A\left(\frac{mT}{n} + \alpha T\right). \tag{21}$$

The real and imaginary parts of Fourier coefficients may be expressed as

$$\mathrm{RE}\{F_n\} = \frac{c_n(0)}{2} \qquad \text{for } n=1 \ldots N \tag{22}$$

and

$$\mathrm{IM}\{F_n\} = -\frac{(-1)^m c_n\left(\frac{1}{2^{k+2}}\right)}{2}, \quad \text{for } k=0,\ldots,\mathrm{trunc}(\log_2 N)-2,\ n=2^k(2m+1), \tag{23}$$

where

$$c_n(\alpha) = \sum_{l=1}^{\mathrm{trunc}(N/n)} \mu(l)\, S(ln,\alpha). \tag{24}$$

The summation in (21) is performed for continuous indices, so in the case of sampled signals the interpolation of values for time instants between samples is necessary. As it was shown in [35], in order to get N spectral coefficients a set of D_N samples is needed

$$D_N = 3\left(\frac{N}{\Pi}\right)^2 + O(N \ln N). \tag{25}$$

For example, over 1500 samples of time signal should be taken to obtain exact 64 complex Fourier coefficients. This rather large number of samples can be reduced by interpolation of analyzed signal, although it creates some distortion of the results.

The interpolation may be done simply by taking nearest neighbouring samples, i.e. zero-order interpolation. For 128-points segment of a sinusoidal signal the maximum error caused by zero-order interpolation is less than 1%. For realization of random Gaussian process the mean square error depends on autocorrelation function (ACF) and number of coefficients [27]. For signals with Gaussian envelope those errors vary from 0.013 to 0.201 of signal power depending on signal bandwidth. For linear interpolation those errors are neglible, but the computing time increases considerably.

The AFT method is well suited for real-time application, especially in parallel structures.

3.3. *Autoregressive modelling* (AR)

The PSD of an autoregressive model of analysed signals (PSD-AR) was calculated according to

$$P_{\mathrm{AR}}(f) = \frac{s^2}{\left|1 + \sum_{k=1}^{p} a(k)\, e^{-j2\Pi fk}\right|^2} \tag{26}$$

where $a(k)$ are model coefficients, p — model order and s^2 — total squared error of the model.

The spectral envelope of the AR model is smooth. This estimator is asymptotically unbiased and for large n (and $N > 2p$) its variance is smaller than for the periodogram [19].

The accuracy of such PSD estimation is limited, due to the limited ability of proper modelling of time series as the autoregressive process. Thus, the PSD-AR of the signal with wideband Gaussian spectral envelope will be biased, also the PSD of noisy signal will be distorted. For narrowband process with a Gaussian spectral envelope the bias is small. In spite of these, AR modelling is widely used in analysis of Doppler signals [18, 37], primary merited by its excellent resolution and good spectral match. It should be noted however, that for low SNR the resolution of the AR spectral estimator is no better than that of the periodogram. Kay [20] concluded, that the effect of injected white noise is to produce a flattened PSD-AR estimate, regardless of the nature of the observed process. In our modelling scheme we have found however, that for Doppler signals (with Gaussian spectral envelope) the prediction error was proportional to SNR, if the model order was optimal (PSD-AR close to true PSD).

We have tested two model identification procedures. First, the model order was determined due to the Akaike Information Criterion (AIC) [2]

$$\mathrm{AIC}(p) = N \ln s + 2p \tag{27}$$

The common method of autoregressive coefficients determination is to estimate autocorelation lags of considered signal and then to solve a set of linear equations called Yule-Walker equations (YWE) [20]. Since the solution of the YWE introduces the smallest amount of peaking in the spectrum of the 'colored noise' [19], therefore we postulate the use of YWE for model order identification of the Doppler signals.

The Singular Value Decomposition (SVD) of the overdetemined YWE (primary selection of model order is greater than expected) may be used for construction of the normalized matrix approximation ratio [11]

$$\upsilon(k) = \frac{\| \mathbf{R}^{(k)} \|}{\| \mathbf{R} \|} = \sqrt{\left(\frac{\sigma_1^2 + \sigma_2^2 + \ldots + \sigma_k^2}{\sigma_1^2 + \sigma_2^2 + \ldots + \sigma_r^2} \right)} \tag{28}$$

where $\mathbf{R}$ — autocorrelation matrix with rank r, $\mathbf{R}^{(k)}$ — submatrix with rank k and σ_i — singular values of $\mathbf{R}$. The ratio $\upsilon(k)$ approaches 1, if k equations completely describes AR model of signal, e.g. for signals without noise. For noisy signals, the construction of overdetermined YWE is rather impossible and the ratio $\upsilon(k)$ doesn't reach unity till $r=k$. Cadzow [11] proved however, that for $\upsilon(k)$ close to unity, $\mathbf{R}^{(k)}$ is the best approximation of $\mathbf{R}$ in the least square sense. Consequently, the selection of the model order was done observing the progression of $\upsilon(k)$ with increasing k. The value of k for which the ratio $\upsilon(k)$ was reaching "plateau" (typically for $\upsilon(k)$ equal to 0.999) was chosen as the AR model order.

Usually, the model order described by SVD was greater than the one obtained by AIC and was used here for model identification only. The difference was observed especially for signals with low SNR and larger bandwidth.

3.4. ARMA *modelling*

The signal described as concatenation of AR and MA processes has the PSD expressed as

$$P_{\mathrm{ARMA}}(f) = \left| \frac{\sum_{k=0}^{q} b(k) e^{-j2\Pi fk}}{\sum_{k=1}^{p} a(k) e^{-j2\Pi fk}} \right| \cdot \tag{29}$$

The ARMA model often provides better spectral estimates than either AR or MA models. Simultaneous evaluation of both a and b parameters in Eq. (29) is computationally ineffective, so the suboptimal solutions should be rather used. We applied the least square solution, based on the Cadzow's [10] least square method (LSMYWE) to tune between better statistical properties and computational cost [20]. The MA coefficients were computed according to Durbin's method [19].

The therоretical considerations concerning variance of the estimator [33] leads to the conclusion, that LSMYWE give asymptotically unbiased estimator with variance monotonically decreasing with increasing number of equations ($N >> p$).

Although numerous techniques were developed, the common solution is to estimate the order of AR process and next to select $q=p$ in (29). This is based on observation of small sensivity of overall ARMA model to inaccuracies in MA identification[1]. The SVD method (applied to the modified YWE) was used for determination of the number of poles p and next the order of MA process was assumed to be equal to the AR order.

4. Mean frequency estimation

The mean frequency was estimated as a first spectral moment of the simulated signal (F1 method):

$$Kv_{\text{mean}} = f_{\text{mean}} = \frac{\sum_{0}^{f_{\max}} fP(f)\,df}{\int_{0}^{f_{\max}} P(f)\,df} \cong \frac{\sum_i f_i P(f_i)}{\sum_i P(f_i)}. \tag{30}$$

Since Eq. (30) estimates the mean of the composite Doppler signal plus noise, so it includes additional bias due to the noise.

One of the simplest method of noise suppression is to subtract the noise spectral density ($N(f)$) from the derived spectrum and then to calculate the mean (F1-NS method):

$$f_{\text{mean(NS)}} = \frac{\sum_{i=0}^{N-1} f_i(P(f_i)-N(f_i))}{\sum_{i=0}^{n-1} (P(f_i)-N(f_i))}. \tag{31}$$

The noise spectral density $N(f)$ was estimated from the tail of the spectrum, beyond maximum frequency od Doppler signal. Sirmans and Bumgarner [31] concluded, that the F1-NS method provided the unbiased estimation of f_{mean} even for low SNRs and that the standard deviation of this estimator was small.

In our study, both, first moment (F1) and first moment with noise suppression (F1-NS) methods were used for mean frequency estimation of all spectral estimators.

5. Maximum frequency estimation

The estimation of $f_{\max}$ is a rather complicated task, resulting from the random nature of Doppler signal and the presence of noise. The misrepresention may occur during the observation of low-level signal in noisy environment.

Since the PSD of the random signal with Gaussian spectral envelope is unbounded, so $f_{\max}$ was defined as the frequency at which spectral power decreased down to 0.1 of its

[1] The pure MA model was rejected after primary analysis due to its poor performance.

maximum value (method 0.1max), although it was realiable only for smoothed PSD (AR, ARMA).

Along with 0.1max two other f_{max} estimators — d'Alessio percentile and modified threshold crossing (MTCM) [23] were examined. It was confirmed, that their behavior strongly depend on SNR, noise level N(f) and the shape of time windowind functoin of the analysed signal.

The novel geometrical method is simpler, being sufficiently precise and computational effective. The integrated power spectrum [23] is equal to

$$\Phi(f) = \int_{f_i}^{f} P(f)\, df. \tag{32}$$

The basic idea of our method is as follows. First the power spectral integral $\Phi(f)$ is computed. Next the straight line connecting two points on $\Phi(f)$ corresponding: 1. to the maximum analysed frequency ($\Phi(f_h)$) and 2. to the value of f_{mode} for which the peak spectral power is maximum ($\Phi(f_{mode})$) is constructed. Then the distance from this line to $\Phi(f)$ is computed for all frequencies greater than f_{mode}. Finally, the frequency at which the distance is maximum is assumed to be the maximum frequency of the signal. On the Fig. 3 an example of $\Phi(f)$ is presented; f_{mode} denotes the frequency where P(f) reaches maximum value, f_h is maximum analysed frequency.

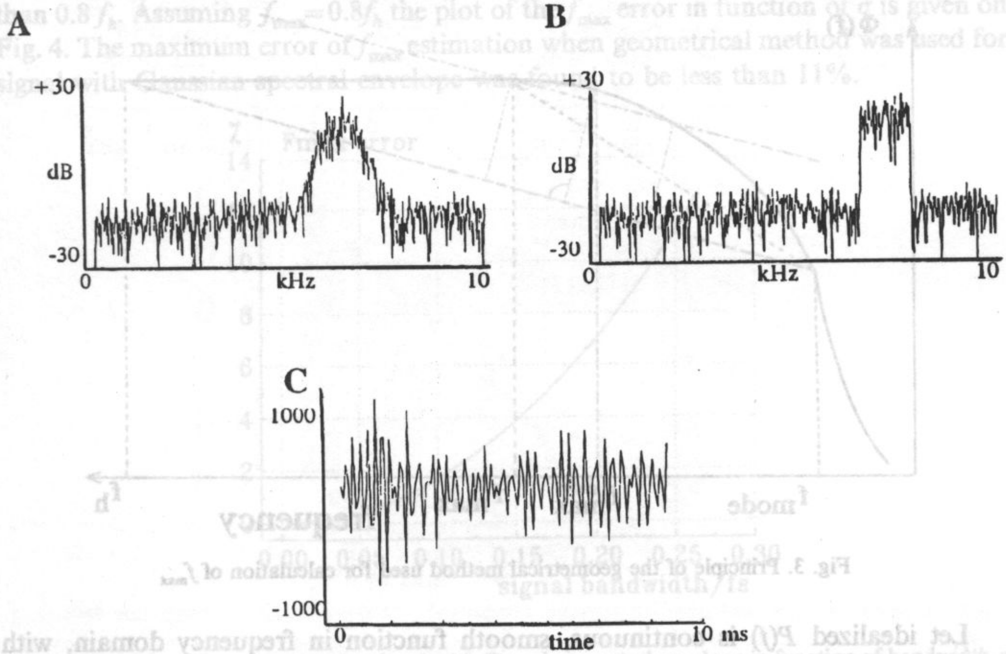


Fig. 1. Simulated spectra, SNR = 20 dB; a) Gaussian envelope, b) rectangular envelope, c) time signal for Gaussian envelope.

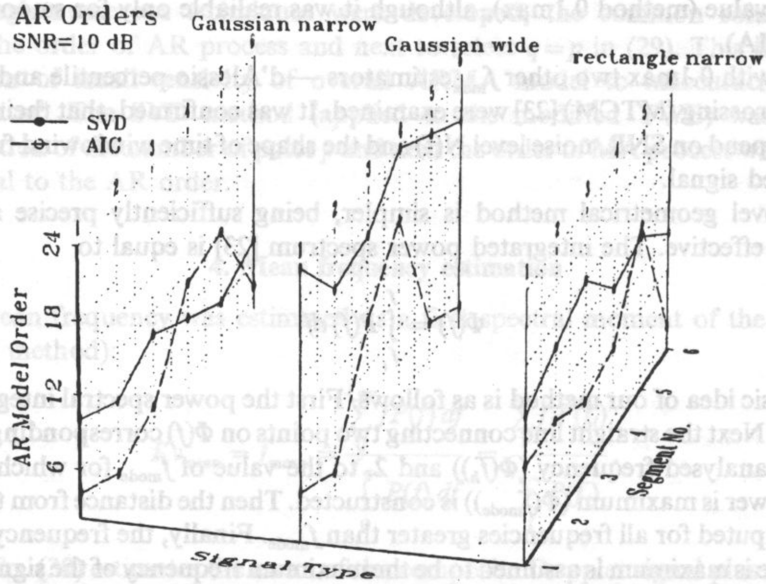


Fig. 2. Examples of AR models orders obtained by AIC and SVD for realizations of the identical processes a) narrow band Gaussian envelope, b) wide band Gaussian envelope, c) rectangular spectrum.

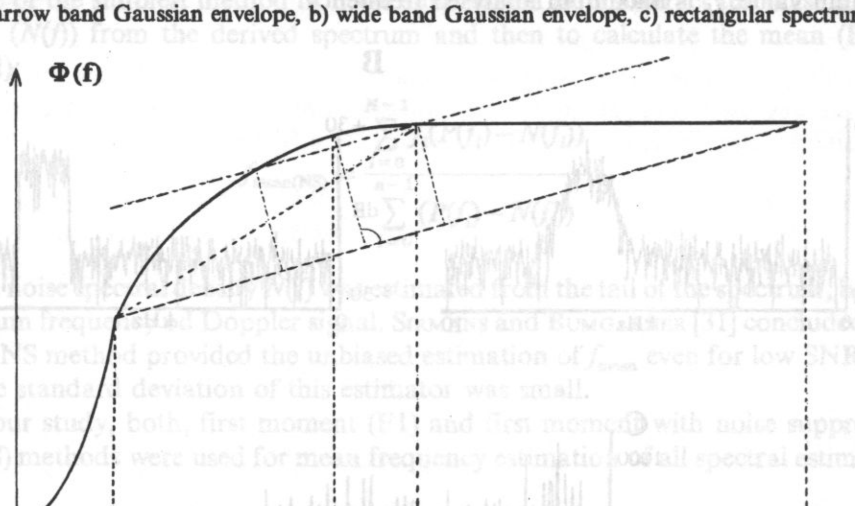


Fig. 3. Principle of the geometrical method used for calculation of f_{max}

Let idealized $P(f)$ is continuous, smooth function in frequency domain, with a single maximum for f_{mode}. Assuming that no noise is present, $P(f) = 0$, for $f > f_{max}$.[2] Let

[2] For averaged spectrum, we can assume that additional noise causes rotation of $\Phi(f)$ relative to axes over the arc of arctan $(N(f))$ radian.

$\Phi(f)$ denotes integral of $P(f)$ according to (32). Then, begining at f_{max} $\Phi(f)$ becomes parallel to f axis and the slope α of the straight line $\Phi(f_{mode})$ $\Phi(f_h)$ is equal to

$$\alpha=\frac{\Phi(f_h)-\Phi(f_{mode})}{f_h-f_{mode}}. \quad (33)$$

The distance d is maximum for $f \in <f_{max}, f_n>$ and such estimation of f_{max} has no positive bias. For frequencies within $<f_{mode}, f_{max}>$ the error of the method depends on proportions between slope of $\Phi(f)$ and slope of the $\Phi(f_{mode})$ $\Phi(f_h)$ line. $D(f)$ is maximum at point $f=f_{max}$ if, and only, if the slope of the line $\Phi(f)\Phi(f_{max})$ is greater then the slope of the line $\Phi(f_{mode})\Phi(f_h)$

$$\forall\,(f_{mode}\leqslant f\leqslant f_{max})\ \frac{\Phi(f_h)-\Phi(f_{mode})}{f_h-f_{mode}} > \frac{\Phi(f_{max})-\Phi(f)}{f_{max}-f}. \quad (34)$$

According to (34): 1. the error is maximum, when $f_{max}=f_h$, 2. the underestimation of f_{max} is decreasing, when $P(f)$ is slowly decreasing for increasing frequency.

Let consider the case of the signal with a Gaussian spectral envelope (Eq. (11)). Its maximum frequency was defined as the frequency at which the spectral envelope decreased to 0.1 of its maximum value. For Gaussian spectral envelope $f_{max}\approx f_{mode}+2.1\sigma$. Practically, the examined Doppler signals extended only to f_{max} less than 0.8 f_h. Assuming $f_{max}=0.8 f_h$ the plot of the f_{max} error in function of σ is given on Fig. 4. The maximum error of f_{max} estimation when geometrical method was used for signal with Gaussian spectral envelope was found to be less than 11%.

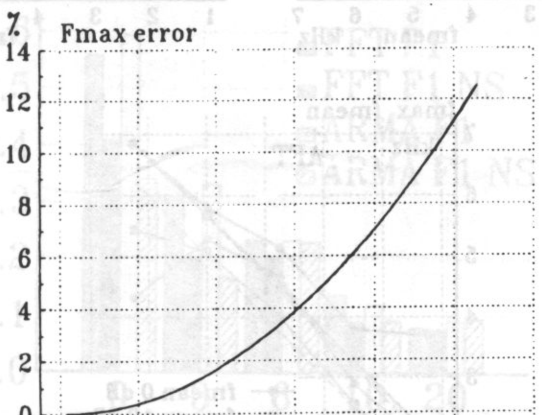


Fig. 4. The error of f_{max} estimation for signal with Gaussian spectral envelope in function of bandwidth σ.

According to Eq. (34) the maximum frequency of signals with rectangular spectral envelope was estimated with no errors.

6. Results and discussion

FFT, AFT, AR Burg and ARMA were tested for PSD estimation of all simulated random processes. F_{mean} was computed using PSD's first moment with and without noise substraction. Finally, f_{max} was computed using 0.1 max method, d'Alessio, threshold crossing method and novel geometrical method. The results were compared to the reference f_{max} selected during simulation at the level of 10% of the peak spectral power. The bias and standard deviation of estimators were computed and compared for every data file (24 data blocks of 128 samples length) for varying SNR (0, 3, 6, 10, 20 dB).

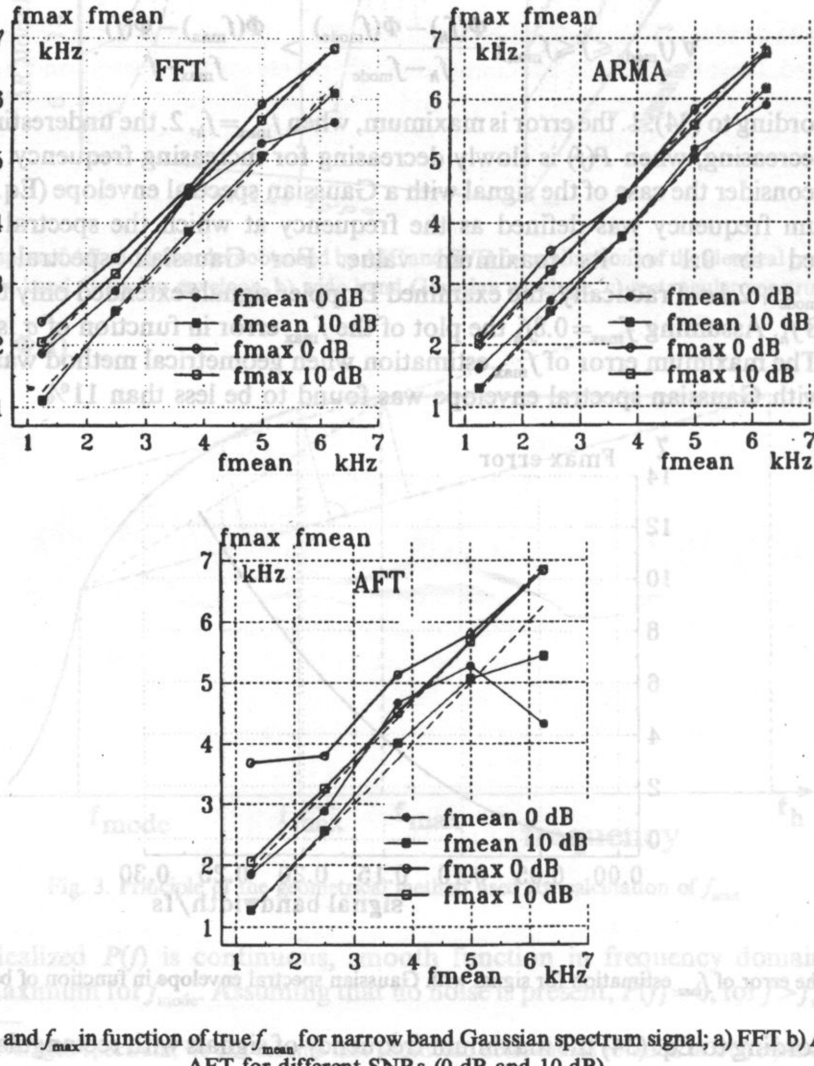
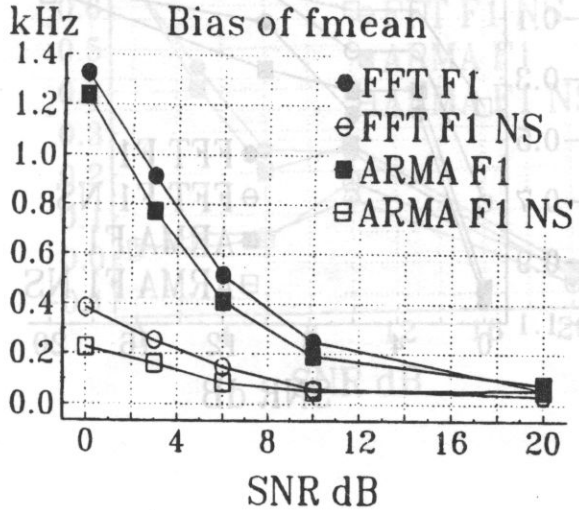


Fig. 5. F_{mean} and f_{max} in function of true f_{mean} for narrow band Gaussian spectrum signal; a) FFT b) ARMA c) AFT for different SNRs (0 dB and 10 dB).

The comparison of different PSD estimations for signal with a narrow-band Gaussian spectral envelope is presented on Fig. 5.



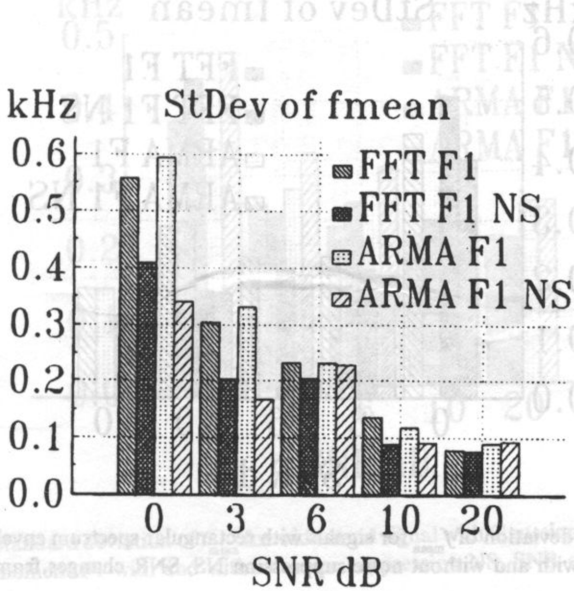


Fig. 6. Bias and standard deviation of f_{mean} for narrow band signal with Gaussian spectrum envelope computed using first moment F1 with and without noise suppression NS, SNR changes from 0 to 20 dB.

The bias of the mean frequency estimation for variable SNRs was lower for the F1-NS method than for F1 method. The standard deviation remained almost unchanged for both, FFT and ARMA methods, (Fig. 6, 7 and 8).

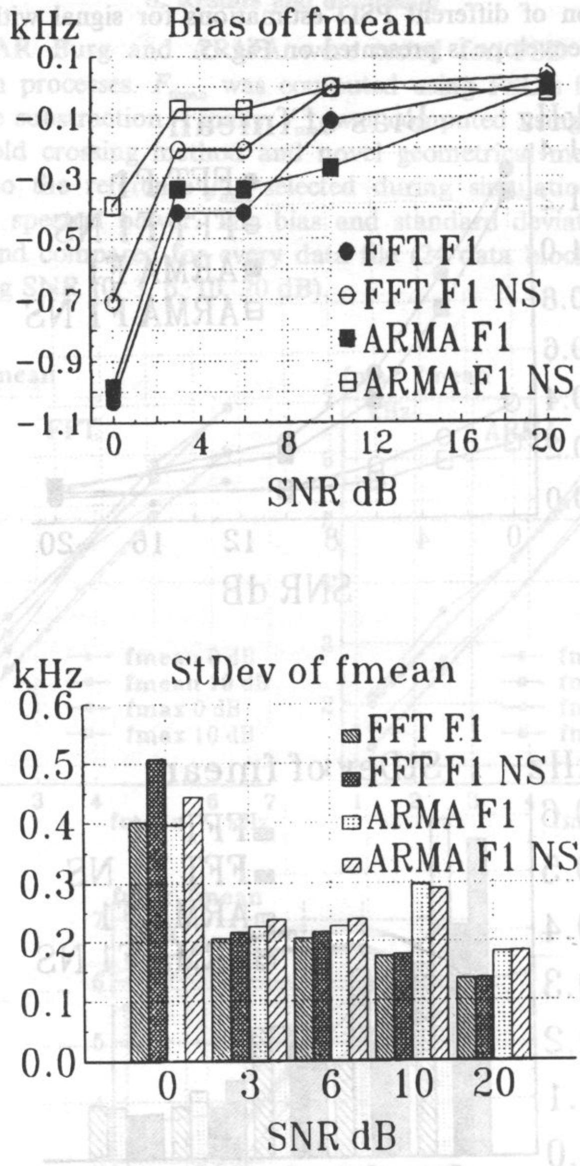


Fig. 7. Bias and standard deviation of f_{mean} for signals with rectangular spectrum envelope computed using first moment F1 with and without noise suppression NS, SNR changes from 0 to 20 dB.

The comparison of f_{max} estimators proved the advantages of the geometrical approach (Fig. 9, 10 and 11) over other tested methods. The bias was significantly reduced, especially for very low SNR's. A reduction in standard deviation was observed showing a significant improvement in performance. Considering the PSD estimation, ARMA modelling was found superior for f_{mean} and f_{max} estimation, while

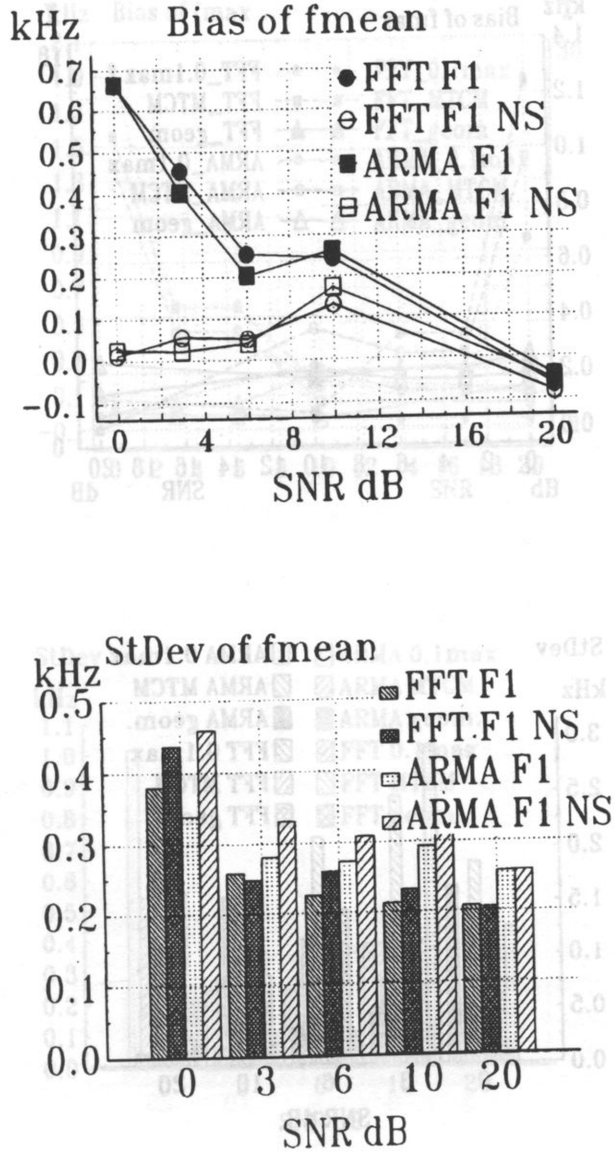
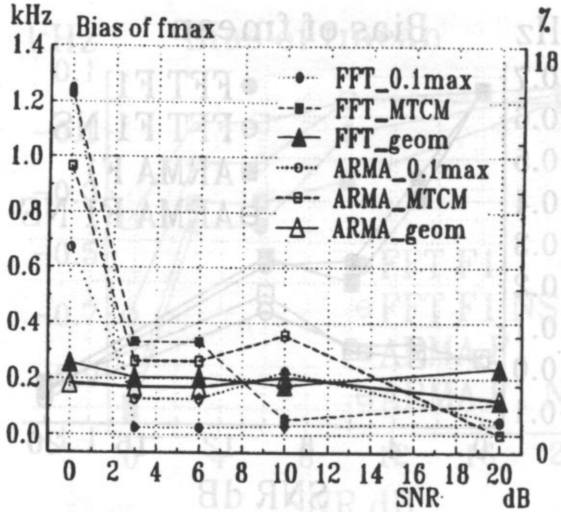


Fig. 8. Bias and standard deviation of f_{mean} for wide-band signal with Gaussian envelope spectrum computed using first moment F1 with and without noise suppression NS, SNR changes from 0 to 20 dB.

the AFT method was apparent worse, especially for low SNR's. The AR model of PSD with order selection after Akaike criterion was found to produce the mean frequency estimation comparable to the one achieved with AR model when order selection was calculated using SVD. The SVD AR performed better for f_{max} estimation.



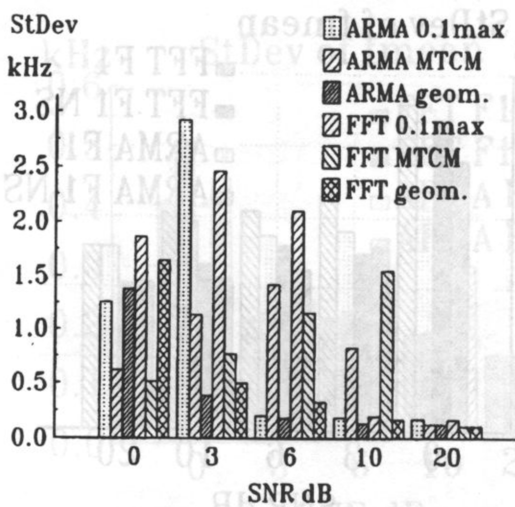
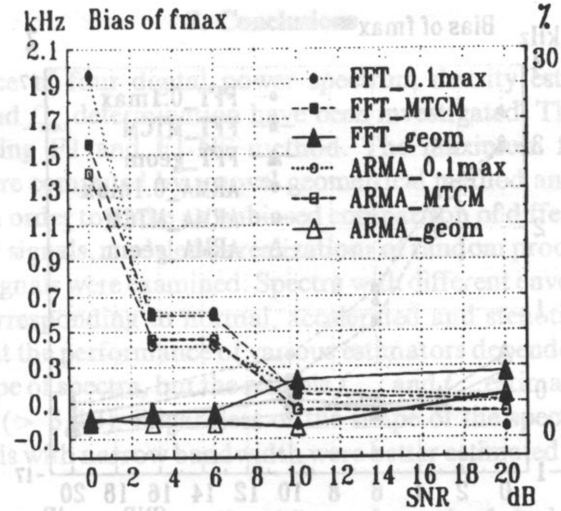


Fig. 9. Bias and standard deviation of f_{max} for narrow band signal with Gaussian envelope spectrum computed using 0.1max, MTCM and geometrical methods, SNR changes from 0 to 20 dB.

The overall performances of both: PSD estimation methods and f_{mean} and f_{max} computations, proved to be similar also for signals with medium and wide band Gaussian envelope, although the bias and standard deviation grew with growing signal bandwidth.



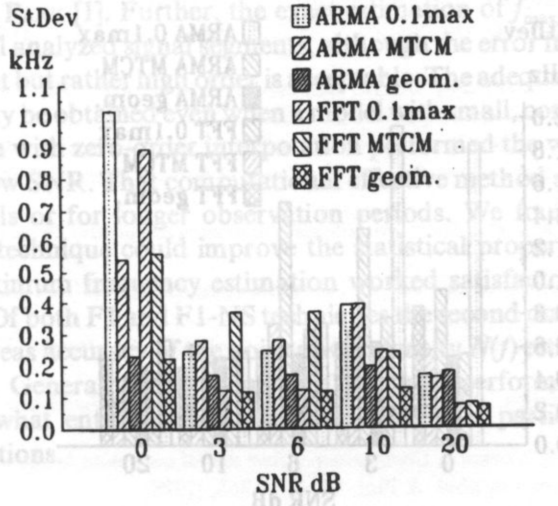
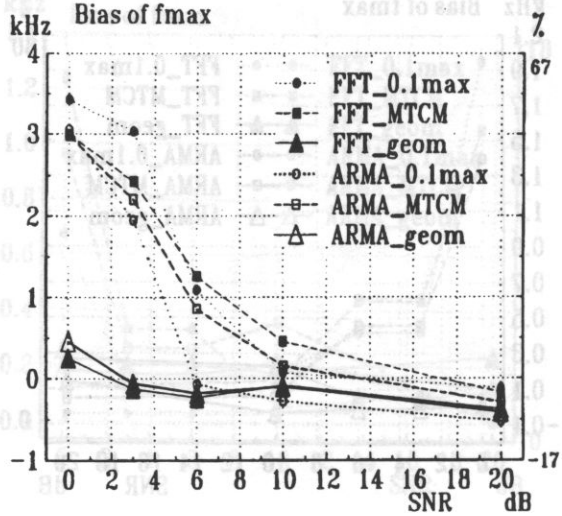


Fig. 10. Bias and standard deviation of f_{max} for signals wigh rectangular spectrum envelope computed using 0.1max, MTCM and geometrical methods, SNR changes from 0 to 20 dB.

For rectangular spectra the differences between the efficiency of estimation of f_{mean} and f_{max} were smaller than for signals with a Gaussian spectral envelope due to more distinct modelling of examined quantities (Fig. 10). The PSDs evaluation has given no unique differences for all methods except AFT. F_{mean} was still better approximated by F1-NS method. The smallest bias and standard deviation of f_{max} were found using the



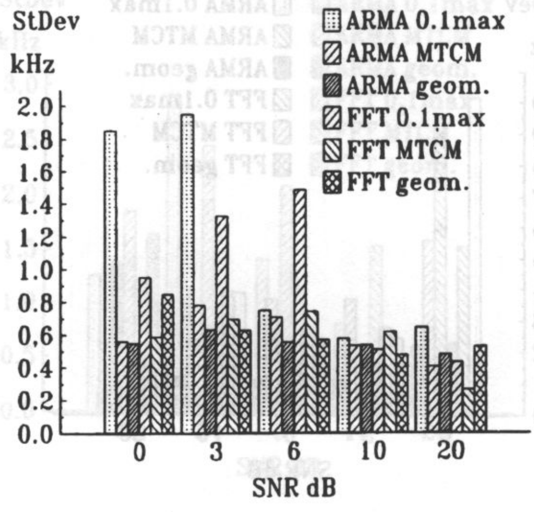


Fig. 11. Bias and standard deviation of f_{max} for the signals with wide-band Gaussian spectrum envelope computed using 0.1max, MTCM and geometrical methods, SNR changes from 0 to 20 dB.

geometrical method. From all other methods of f_{max} estimation the MTCM performed the best, but was comparable to geometrical one only for SNRs > 6 dB. The location of f_{mode} didn't affect the results. Generally, the bias and standard deviation of the results were increasing for wider spectra (Fig. 9 and 11).

7. Conclusions

The performance of four digital power spectrum density estimators and their influence on f_{mean} and f_{max} determination have been investigated. The mean frequency was determined using F1 and F1-NS method. The maximum frequencies of the Doppler spectra were computed by a novel geometrical method and compared to the other techniques. In order to make an unbiased comparison of different estimators the computer simulated signals, modeled as realizations of random processes equivalent to real Doppler flow signals were examined. Spectra with different envelopes were used to simulate signals corresponding to normal, accelerated and stenotic blood flows.

It was found that the performance of various estimators depended on SNRs, signal bandwidth and shape of spectra, but the reliable f_{mean} and f_{max} estimation were obtained also for low SNRs (> 3 dB). Regardless of the shape of the spectrum and the PSD estimator, the signals with narrow bandwidth were better estimated than the wideband signals.

ARMA model displayed the smallest bias and standard deviation of mean and maximum frequency estimates. Also autoregressive models (especially with the model order predicted by SVD) performed better than FFT. In general, exact model identification influenced mor the f_{max} estimation rather than f_{mean}, confirming the recent results of Ahn and Park [1]. Further, the exact estimation of f_{max} needs model order identification for all analyzed signal segments, although the error introduced using the model with constant but rather high order is acceptable. The adequate estimation of the mean frequency may be obtained even when a model with small, constant order is used.

AFT estimation with zero-order interpolation performed the worse, especially for signals with very low SNR. That computational effective method should be used only for "strong" signals or for longer observation periods. We found, that the use of "sliding window" technique could improve the statistical properties of AFT.

Mean and maximum frequency estimation worked satisfactory only for signals with SNR > 3 dB. Of both F1 and F1-NS techniques the second one was giving usually better results, whereas accuracy of the noise power density $N(f)$ estimation was limited [Mo et al., 1988]. Generally, the geometrical method performed the best for all analyzed signals, what entitle us to conclusion, that it is particularly suitable for real-time computations.

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REFLECTION COEFFICIENT FOR TWO ELASTIC LAYERS JOINING TWO HOMOGENEOUS MATERIALS

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Two homogeneous elastic layers are situated between two homogeneous elastic materials. The reflection coefficient for the harmonic wave depends on the elastic constants of the layers and the frequency. The formula is too complex for an analytical treatment. Two situations were analysed numerically. In the first one, thicknesses of the layers were kept constant, and the speeds leading to constant reflection coefficient were calculated. In this case the reflection coefficient either has no minimum, or its minimum equals zero. In the second situation, propagation speeds were constant, and the thickness leading to constant reflection coefficient were calculated. There exist minima equal to zero, and maxima equal to the reflection coefficient for the long-wave limit.

1. Introduction

Between two adjoining homogeneous materials usually there exists a transition zone. The incident harmonic wave arriving at the transition zone splits into the reflected and the transmitted wave. The ratio of the energy flux of the reflected wave to the energy flux of incident wave is the reflection coefficient. There exists no tool for analytic optimisation of the general continuous transition from one to the other propagation speed. In this paper the transition region is approximated by two homogeneous elastic layers. The analysis of the reflection coefficient for this situation is given. One interesting qualitative result is obtained.

2. Jump discontinuities

In general, the transition zone between two adjoining materials, due to technology (e.g. welding, glueing) is inhomogeneous. The reflection coefficient λ for such situation is a functional of the function $c(x)$, where c is the wave speed and x the distance. It is easy to write the corresponding equations, and calculate λ for a $c(x)$ given in advance. In numerous situations the analytical formula may be obtained, cf. e.g. [1]. The only difficulty is connected with finding the solutions of an ordinary differential equation with variable coefficients. For λ given in advance many different $c(x)$ may be calculated. It is impossible, however, to find $c(x)$ leading to minimum of λ, since it is impossible to write λ as the functional of $c(x)$. This is due to the fact, that it is impossible to write explicitly the solutions of the ordinary differential equation as the function of its coefficients.

Because of this difficulty, the inhomogeneous transition zone in this paper is approximated by two homogeneous layers. Already for this very simple model interesting qualitative results are obtained. Each of the four materials considered (two fixed half-space and two layers) is identified by the subscripts 0, 1, 2, 3 (Fig. 1).

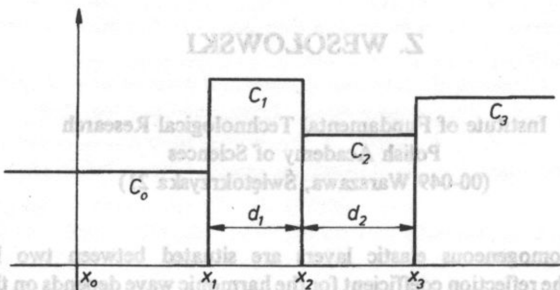


FIG. 1

The harmonic waves propagate in the direction perpendicular to the layer and the displacement u in the k-th material consists of two sinusoidal waves, the first running to the right and the second running to the left,

$$u = A_k \exp i\omega \left[t - \frac{x - x_k}{c_k} \right] + B_k \exp i\omega \left[t + \frac{x - x_k}{c_k} \right]. \tag{1.1}$$

At the boundaries between the layers both the displacement and stress are continuous. It follows that the amplitudes A_k, B_k are connected by the matrix relations (cf. e.g. [2])

$$\begin{bmatrix} A_k \\ B_k \end{bmatrix} = M_k \begin{bmatrix} A_{k-1} \\ B_{k-1} \end{bmatrix}, \tag{1.2}$$

where

$$M_k = \begin{bmatrix} (1+\varkappa_k)\exp(-i\alpha_k) & (1-\varkappa_k)\exp(i\alpha_k) \\ (1-\varkappa_k)\exp(-i\alpha_k) & (1+\varkappa_k)\exp(i\alpha_k) \end{bmatrix}. \tag{1.3}$$

$$\varkappa_k = \frac{\rho_{k-1} c_{k-1}}{\rho_k c_k}, \qquad \alpha_k = \omega \frac{x_k - x_{k-1}}{c_{k-1}}, \tag{1.4}$$

and ρ_k is the density. The transition matrix M_k is non-singular, therefore always its inverse M_k^{-1} exists. Chaining the formulae (1.2) for subsequent $K = 1, 2, 3$, the amplitudes A_3, B_3 may be expressed by A_0, B_0, and vice versa,

$$\begin{bmatrix} A_3 \\ B_3 \end{bmatrix} = M_3\, M_2\, M_1 \begin{bmatrix} A_0 \\ B_0 \end{bmatrix}, \quad \begin{bmatrix} A_0 \\ B_0 \end{bmatrix} = M_1^{-1}\, M_2^{-1}\, M_3^{-1} \begin{bmatrix} A_3 \\ B_3 \end{bmatrix}. \tag{1.5}$$

It is seen that two amplitudes may be taken at will. If we take $B_3 = 0$ and prescribe the value of A_0, then A_0, A_3, B_3 represent the amplitudes of the incident wave, the transmitted wave (both running to the right), and the reflected wave (running to the left). If we take $A_0 = 0$, then B_3, B_0, A_0 represent the amplitudes of the incident an the reflected waves both running to the left and the reflected wave running to the right.

In our problem the speeds c_0, c_3 are given in advance and the speeds c_1, c_2 are free parameters. The densities ρ_k are assumed to be equal to each other. No difficulty is connected with taking into account different densities. The positions x_k will be defined when performing the numerical calculations.

Now we take $A_0 = 0$ and consider the term proportional to B_n as the incident wave, and the terms proportional to A_n, B_0 as the reflected and transmitted waves, respectively. The other possible choice $B_3 = 0$ leads to the same reflection coefficient [1], since the system of layers has no directional properties.

In accord with the above relations, the following expressions for A_3, B_3 are obtained

$$8A_3 = B_0 \exp(-\alpha_1) \times \tag{1.6}$$

$$\times [(1-\varkappa_1)(1+\varkappa_2)(1+\varkappa_3)\exp(+\alpha_2+\alpha_3) + (1-\varkappa_1)(1-\varkappa_2)(1-\varkappa_3)\exp(+\alpha_2-\alpha_3) +$$
$$+ (1+\varkappa_1)(1-\varkappa_2)(1+\varkappa_3)\exp(-\alpha_2+\alpha_3) + (1+\varkappa_1)(1+\varkappa_2)(1-\varkappa_3)\exp(-\alpha_2-\alpha_3)].$$

$$8B_3 = B_0 \exp(-\alpha_1) \times \tag{1.7}$$

$$\times [(1-\varkappa_1)(1+\varkappa_2)(1-\varkappa_3)\exp(+\alpha_2+\alpha_3) + (1-\varkappa_1)(1-\varkappa_2)(1+\varkappa_3)\exp(+\alpha_2-\alpha_3) +$$
$$+ (1+\varkappa_1)(1-\varkappa_2)(1-\varkappa_3)\exp(-\alpha_2+\alpha_3) + (1+\varkappa_1)(1+\varkappa_2)(1+\varkappa_3)\exp(-\alpha_2-\alpha_3)].$$

Without restricting the generality in further calculations we assume $\alpha_1 = 0$.

The right-hand sides of (1.6), (1.7) are complex numbers. Their squared moduli are given by the formulae

$$64 A_3 \overline{A_3} = B_0 \overline{B_0}\, [D_1^2 + D_2^2 + D_3^2 + D_4^2 + 2\,(D_1 D_3 + D_2 D_4) \cos 2\alpha_2 + \tag{1.8}$$
$$+ 2\,(D_1 D_2 + D_3 D_4) \cos 2\alpha_3 + 2\, D_1 D_4 \cos (2\alpha_2 + 2\alpha_3) +$$
$$+ 2\, D_2 D_3 \cos (2\alpha_2 - 2\alpha_3)],$$

$$64 B_3 \overline{B_3} = B_0 \overline{B_0}\, [D_5^2 + D_6^2 + D_7^2 + D_8^2 + 2\,(D_5 D_7 + D_6 D_8) \cos 2\alpha_2 + \tag{1.9}$$
$$+ 2\,(D_5 D_6 + D_7 D_8) \cos 2\alpha_3 + 2\, D_5 D_8 \cos (2\alpha_2 + 2\alpha_3) +$$
$$+ 2\, D_6 D_7 \cos (2\alpha_2 - 2\alpha_3)],$$

where the coefficients D_1 depend only on the speed ratios $\varkappa_k$.

$$D_1 = (1-\varkappa_1)(1+\varkappa_2)(1+\varkappa_3), \quad D_2 = (1-\varkappa_1)(1-\varkappa_2)(1-\varkappa_3),$$
$$D_3 = (1+\varkappa_1)(1-\varkappa_2)(1+\varkappa_3), \quad D_4 = (1+\varkappa_1)(1+\varkappa_2)(1-\varkappa_3), \tag{1.10}$$
$$D_5 = (1-\varkappa_1)(1+\varkappa_2)(1-\varkappa_3), \quad D_6 = (1-\varkappa_1)(1-\varkappa_2)(1+\varkappa_3),$$
$$D_7 = (1+\varkappa_1)(1-\varkappa_2)(1-\varkappa_3), \quad D_8 = (1+\varkappa_1)(1+\varkappa_2)(1+\varkappa_3),$$

Energy flux corresponding to the wave of amplitude A_3 and speed c_3 equals $A_3\overline{A_3}/c_3$. Analogous relations hold for the remaining waves. The reflection coefficient λ equals the ratio of the reflected energy flux to the incident energy flux. Therefore

$$\lambda = \frac{A_3\overline{A_3}}{B_3\overline{B_3}}. \tag{1.11}$$

Obviously $0<\lambda<1$. The first inequality follows from (1.11), since both the nominator and denominator are positive, and the second follows from the energy conservation law.

We write explicity the complete formula for resulting from substitution of (1.7)–(1.9) into (1.10). We obtain

$$\begin{aligned}\lambda = {} & [D_1^2+D_2^2+D_3^2+D_4^2+2(D_1D_3+D_2D_4)\cos 2\alpha_2+ \\ & +2(D_1D_2+D_3D_4)\cos 2\alpha_3+2D_1D_4\cos(2\alpha_2+2\alpha_3)+ \\ & +2D_2D_3\cos(2\alpha_2-2\alpha_3)]\times \\ & \times[D_5^2+D_6^2+D_7^2+D_8^2+2(D_5D_7+D_6D_8)\cos 2\alpha_2+ \\ & +2(D_5D_6+D_7D_8)\cos 2\alpha_3+2D_5D_8\cos(2\alpha_2+2\alpha_3)+ \\ & +2D_6D_7\cos(2\alpha_2-2\alpha_3)]^{-1},\end{aligned} \tag{1.12}$$

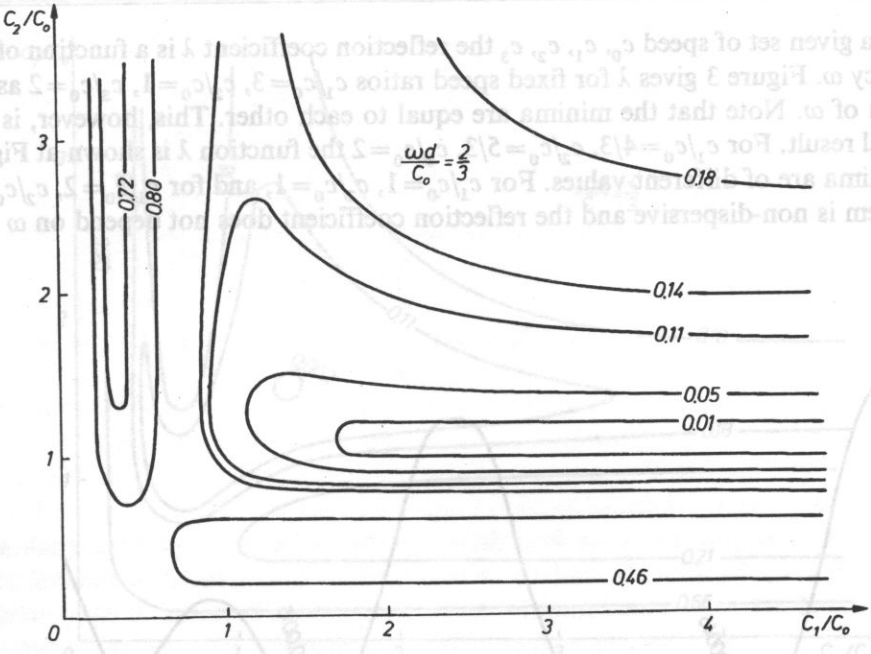
where D_K are defined by (1.10).

In order to find the extremum value of λ, the derivatives of the function (1.11) with respect to c_1 and c_2 must be calculated and put equal to zero. Note that D_K are functions of c_1, c_2, and therefore the corresponding system of trigonometric equations is very complex and no satisfactory analytic treatment of the equations may be expected. Therefore, we are forced to base on the numerical approach.

2. Fixed thickness, variable speeds

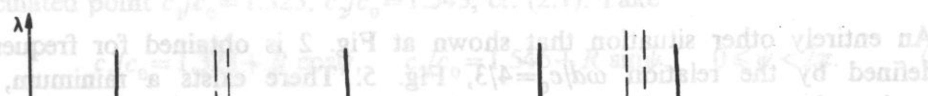
We intend to analyse in this chapter the value of the reflection coefficient, as a function of the two propagation speeds in the layers. Thickness of the layers are kept constant.

Assume ρ_k = const., $x_0=x_1=0$, $x_2=d$, $x_3=2d$. The speed ratio for the homogeneous materials is assumed to be equal two, $c_3=2c_0$. The ratios c_1/c_0 and c_2/c_0 are the two independent variables. Figure 2 gives the curves of constant λ for fixed $\omega d/c_0=2/3$. It is seen that no minimum exists for c_1/c_0, $c_2/c_0<4$. Numerical analysis for larger speeds c_1, c_2 proves that also in other intervals there exists no minimum. Note that only the minimum value is interestings. The maximum value $\lambda=1$ may be reached in the trivial reflection from the free end of from the rigid support.



C2/C0
3
2
1
0
1
2
3
4
C1/C0
0.72
0.80
ωd/C0 = 2/3
0.18
0.14
0.11
0.05
0.01
0.46

FIG. 2



λ
0.1
0.1111
0.1111
0.004762
0.004762
0.004762
0.004762
0
1
8
9
10
18
19
ω

FIG. 3

For a given set of speed c_0, c_1, c_2, c_3 the reflection coefficient λ is a function of the frequency ω. Figure 3 gives λ for fixed speed ratios $c_1/c_0=3$, $c_2/c_0=1$, $c_3/c_0=2$ as the function of ω. Note that the minima are equal to each other. This, however, is not a general result. For $c_1/c_0=4/3$, $c_2/c_0=5/3$, $c_3/c_0=2$ the function λ is shown at Fig. 4. The minima are of different values. For $c_1/c_0=1$, $c_2/c_0=1$, and for $c_1/c_0=2$, $c_2/c_0=2$ the system is non-dispersive and the reflection coefficient does not depend on ω and $\lambda=1/9$.

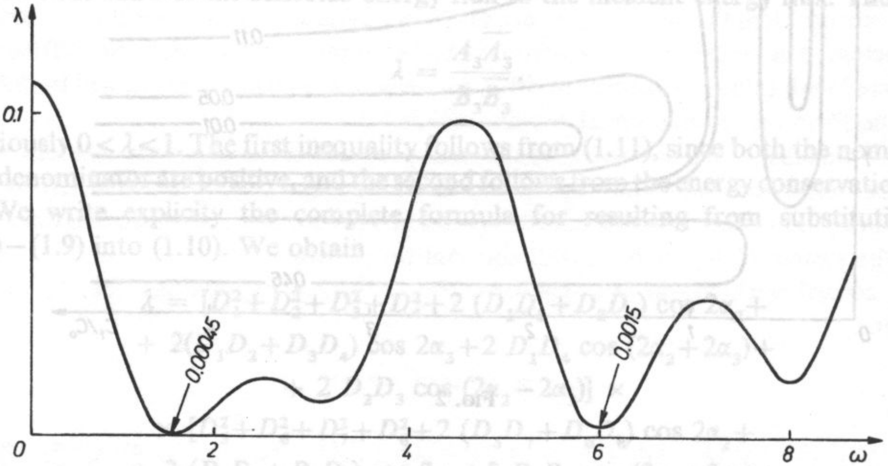


FIG. 4

An entirely other situation that shown at Fig. 2 is obtained for frequency ω defined by the relation $\omega d/c_0=4/3$, Fig. 5. There exists a minimum, at approximately

$$c_1/c_0=1.323, \quad c_2/c_0=1.545. \tag{2.1}$$

The value of the minimum was calculated to be smaller than 10^{-7}. Note that $\lambda=0$ would mean that the structure for the assumed frequency ω is perfectly transparent. Because of this important qualitative result we would like to know whether the minimum value of λ is exactly equal zero.

In order to decide whether $\lambda_{\min}=0$, let's instead of λ consider the values of A_3 near the point $c_1/c_0=1.323$, $c_2/c_0=1.545$. In accord with (1.6) there is

$$8\,\mathrm{Re}\,A_3 = [(1-\varkappa_1)(1+\varkappa_2)(1+\varkappa_3)+(1+\varkappa_1)(1+\varkappa_2)(1-\varkappa_3)]\cos(\alpha_2+\alpha_3)+ \\ +[(1-\varkappa_1)(1-\varkappa_2)(1-\varkappa_3)+(1+\varkappa_1)(1-\varkappa_2)(1+\varkappa_3)]\cos(\alpha_2-\alpha_3). \tag{2.2}$$

$$8\,\mathrm{Im}\,A_3 = [(1-\varkappa_1)(1+\varkappa_2)(1+\varkappa_3)-(1+\varkappa_1)(1+\varkappa_2)(1-\varkappa_3)]\sin(\alpha_2+\alpha_3)+ \\ +[(1-\varkappa_1)(1-\varkappa_2)(1-\varkappa_3)-(1+\varkappa_1)(1-\varkappa_2)(1+\varkappa_3)]\sin(\alpha_2-\alpha_3). \tag{2.3}$$

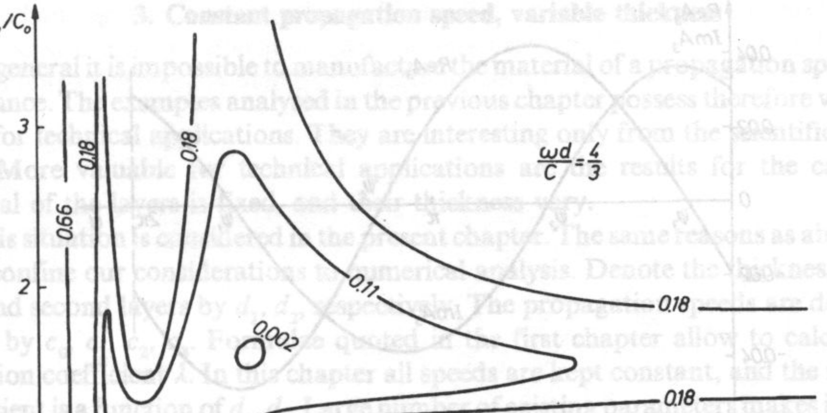


FIG. 5

Consider the values of the complex amplitude A_3 on a circle surrounding the above calculated point $c_1/c_0=1.323$, $c_2/c_0=1.545$, cf. (2.1). Take

$$c_1/c_0=1.323+R\cos\psi,\quad c_2/c_0=1.545+R\sin\psi,\quad 0\leqslant\psi<2\pi. \tag{2.4}$$

where the radius R = const and ψ is a variable parameter, and calculate Re A_3 and Im A_3 as a function of ψ. At the Fig. 6 are given the values of Re A_3 and Im A_3 for $R=1$ as the function of ψ. For $\psi<\psi_1$ and $\psi>\psi_2$ there is Re $A_3>0$, and for $\psi_3<\psi<\psi_4$ there is Im $A_3>0$. In the remaining intervals they are negative or zero. Evidently Re A_3 and Im A_3 are continuous functions of c_1, c_2. Fig. 6 proves that on the plane c_1, c_2 there are four regions where Re A_3 and Im A_3 are i) both positive, ii) positive and negative, iii) negative and positive, and finally *iv*) both negative, Fig. 7. It follows that inside the considered circle there exists a point K, for which there is

$$\text{Re } A_3 = \text{Im } A_3 = 0,\ \lambda_{\min} = 0. \tag{2.5}$$

For completeness the function $\lambda(\omega)$ for the values (2.1) was calculated and sketched at Fig. 8. The minimum value $\omega = 0$ is reached for $\omega=4/3$.

For $\omega d/c_0=8/3$ the minimum corresponds to $c_1/c_0=1.594$, $c_2/c_0=2.290$ and equals zero, too. For other dimensions, materials and frequencies the obtained figures were always similar either to Fig. 2 or similar to Fig. 3. If a map similar to that given at Fig. 3 was obtained, then the minimum would be equal to zero.

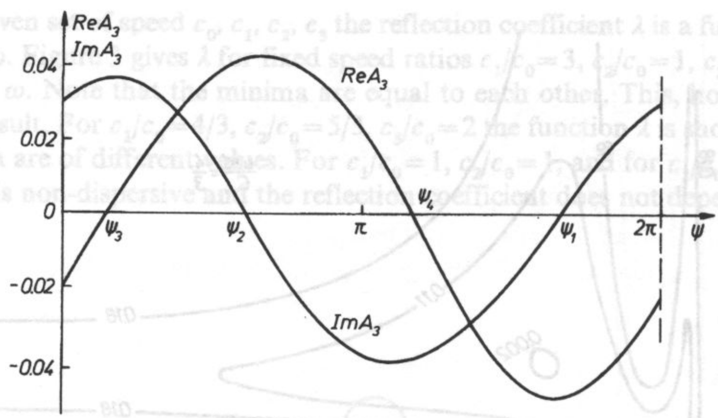


FIG. 6

For two transition layers we face therefore the following important qualitative result for a given in advance and fixed frequency ω and thickness: the reflection coefficient λ either has no minimum for finite c_1, c_2 or has a minimum equal to zero. Since the result was obtained numerically, its generality may be restricted. No satisfactory physical explanation of the fact that the minimum equals zero is known to the author.

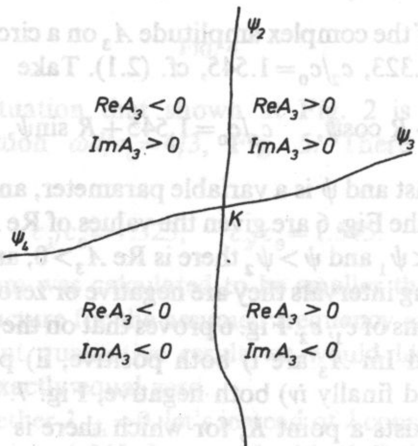


FIG. 7

For larger number of layers the above qualitative result was numerically checked in few numerical examples. It seems that the following qualitative result holds: either there exist no minimum for finite speeds, ot there exists the minimum equal to zero. This result demands further analysis. It is not known, if it is general.

3. Constant propagation speed, variable thickness

In general it is impossible to manufacture the material of a propagation speed given in advance. The examples analysed in the previous chapter possess therefore very small value for technical applications. They are interesting only from the scientific point of view. More valuable for technical applications are the results for the case when material of the layers is fixed, and their thickness vary.

This situation is considered in the present chapter. The same reasons as above force us to confine our considerations to numerical analysis. Denote the thicknesses of the first and second layers by d_1, d_2, respectively. The propagation speeds are denoted as above by c_0, c_1, c_2, c_3. Formulae quoted in the first chapter allow to calculate the reflection coefficient λ. In this chapter all speeds are kept constant, and the reflection coefficient is a function of d_1, d_2. Large number of existing parameters makes it difficult to exhaust all possibilities. Here we consider one example only.

Take $c_0=1$, $c_1=1.4$, $c_2=1.6$, $c_3=2$, $\omega=1$ in some length and time units. It is easy to introduce the dimensionless variables, but because we intend to give only illustrative examples, we prefer to leave the data in the form as above. Figure 8 gives the curves of constant reflection coefficient for $d_1<3$, $d_2<2$. In this interval there exists no extremum.

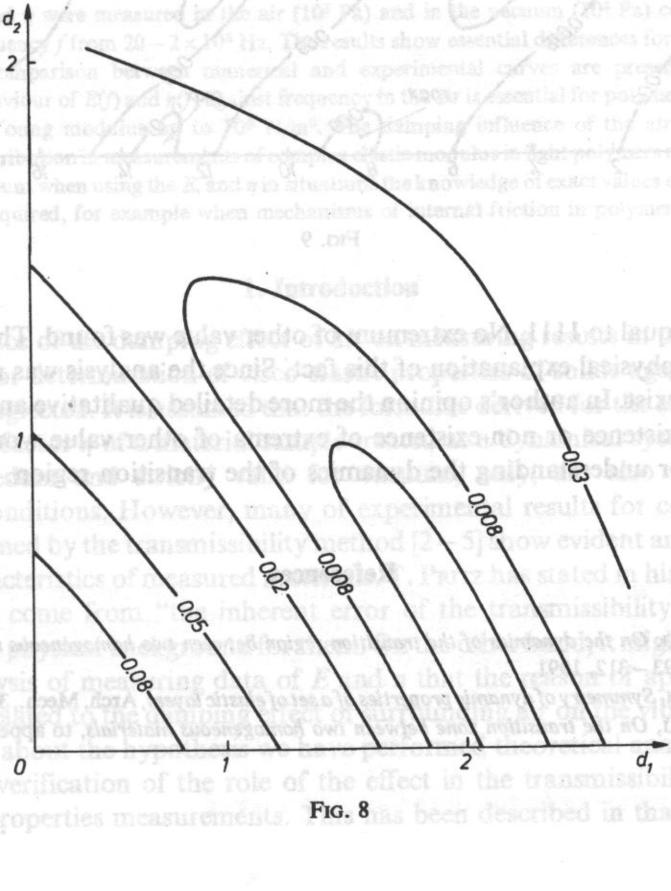


FIG. 8

For large thicknesses the picture is very irregular, since for large thickness the motion of the transition region dominates the behaviour of the system. Figure 9 gives some curves of the constant reflection coefficient λ. There are minima equal to zero,

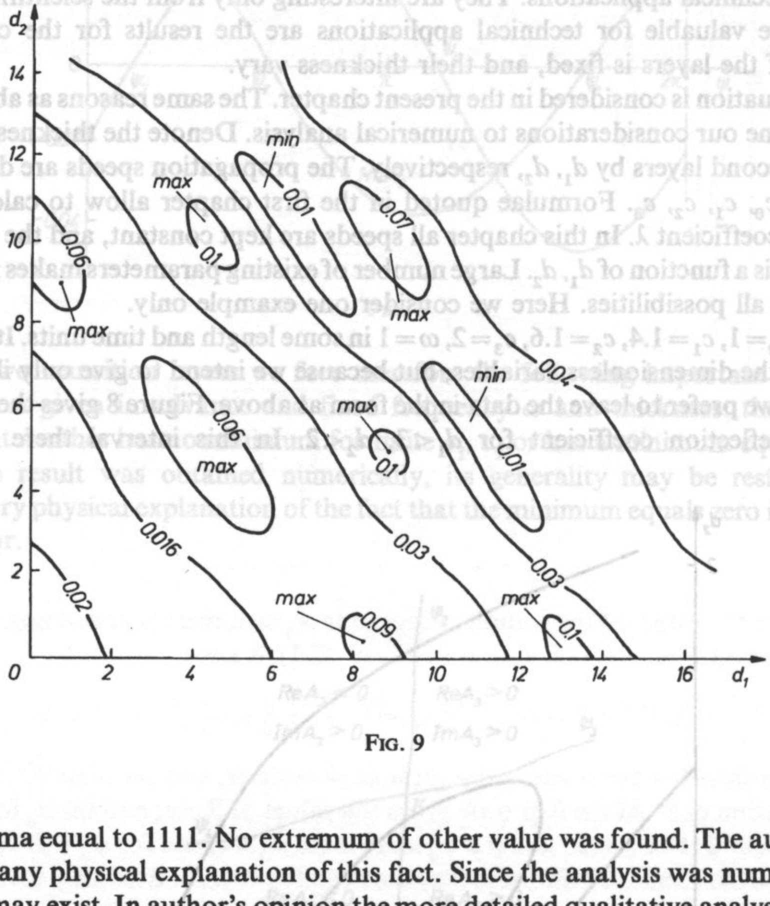


FIG. 9

and maxima equal to 1111. No extremum of other value was found. The author is not aware of any physical explanation of this fact. Since the analysis was numerical such extrema may exist. In author's opinion the more detailed qualitative analysis aimed at proving the existence or non-existence of extrema of other value would be of large importance for understanding the dynamics of the transition region.

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A ROLE OF AIR IN COMPLEX ELASTIC MODULUS MEASUREMENTS OF SOLID SAMPLES BY TRANSMISSIBILITY METHOD

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Anomalous dependence of visco-elastic parameters against frequency for solid polymers measured both in conditions of surrounding air as well as in vacuum were examined. In theoretical consideration the single system of one degree of freedom for representing a sample was assumed together with the presence of friction force introduced by the air. The values of E and η were measured in the air (10^5 Pa) and in the vacuum (10^2 Pa) conditions against frequency f from $20-2\times10^3$ Hz. The results show essential differences for both conditions. A comparison between numerical and experimental curves are presented. Anomalous behaviour of $E(f)$ and $\eta(f)$ against frequency in the air is essential for polymers of small values of Young modulus up to 10^6 N/m². The damping influence of the air having essential contribution in measurements of complex elastic modulus in light polymers must be taken into account when using the E, and η in situations the knowledge of exact values of those quantities is required, for example when mechanisms of internal friction in polymers are evaluated.

1. Introduction

The influence of the damping effect of air on measuring results in transmissibility method used for determination of visco-elastic properties of solids against frequency [1] is usually neglected. It is assumed that the formulae derived for the elastic modulus E and the loss factor η of a material sample treated as a dynamical system of a single degree of freedom and strictly valid for vacuum, only, are also correct in air atmosphere conditions. However, many of experimental results for complex elastic modulus obtained by the transmissibility method [2–5] show evident anomalies in the dynamic characteristics of measured materials. T. Pritz has stated in his paper [3] that the anomalies come from "the inherent error of the transmissibility method" not explaining any physical background for them. On the other hand, it might be suspected from the analysis of measuring data of E and η that the reason of appearing of the anomalies is related to the damping effect of surrounding air on the vibrating sample. To make sure about the hypothesis we have performed theoretical analysis as well as experimental verification of the role of the effect in the transmissibility method of visco-elastic properties measurements. This has been described in that paper.

Most of the authors [3 – 7] obtained their results by measuring the transmissibility T and the phase angle ϕ for different materials in the air but applying determination of them the formulae valid in vacuum, only, so their dynamical characteristics are significantly charged by the systematic error.

There are, also other sources of systematic errors in the method coming from damping of the glue used for cementing the specimen to the shaker, of the mass effect of an accelerometer [3 – 8] and of a cable, however these influences were neglected in the theoretical consideration given below. G.W. LAIRD and H.B. KINGSBURY [4] described how these errors can be eliminated. The glue effect may be neglected when its dynamical characteristics (E, η) are close to the ones of the polymer specimen being examined.

2. The complex transmissibility of a linear single degree of freedom system

A linear system of one degree of freedom is characterized by one resonant frequency and comprises of a single element of mass and one or few elements of stiffness and damping. Example of such single system may be a sample of a visco-elastic material of dimensions of much smaller than the corresponding to the vibration frequency wavelength in the material.

The Fig. 1 presents an element of mass supported by a visco-elastic sample laying on a foundation which vibrates sinusoidally with the frequency $f = \omega/2\pi$. It is assumed that the mass is supported at its center of gravity. The system is situated in the surrounding air which contributes an extraneous damping in its vibrations movement.

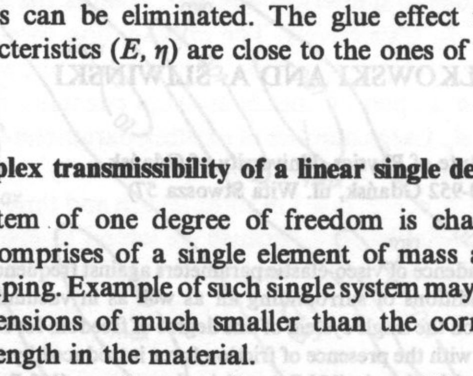

Fig. 1. The simple system representing the sample and loading mass.

This additional damping may be avoid by putting the system into the vacuum however it introduces in practice additional difficulties. Let F_0 represents the force of the air friction. For a small displacement of the mass M the force F_0 may be treated as a linear function of vibrational velocity, i.e. of the frequency as follows

$$F_0 = -\beta\,\dot{\tilde{x}}(t) = -\beta\omega\,\tilde{x}(t), \tag{1}$$

where β is the damping factor of the air and $\dot{\tilde{x}} = i\omega\tilde{x}$.

For greater vibrational velocities (but still less than the velocity of sound in the air) the force F_0 is a function of $\dot{x}^2$, however we shall consider only the linear case (1), here.

The equation of motion for the system may be written as

$$M\frac{d^2\tilde{x}_2(t)}{dt^2} = KE^*(\tilde{x}_1-\tilde{x}_2)-\beta(\dot{\tilde{x}}_1-\dot{\tilde{x}}_2) = KE^*\tilde{x}_1-KE^*\tilde{x}_2+\beta\dot{\tilde{x}}_2-\beta\dot{\tilde{x}}_1, \quad (2)$$

where the constant K is determined by the formula [6]:

$$K = (3A/L)\,(1+bs^2), \quad (3)$$

b is a numerical constant and for a rubber-like samples is equal 2, S is equal to the ratio of the loaded surface to the total force-free area. This is so called a shape factor and for rubber-like materials, for example for a cylinder of the diameter D and the height L the S is equal to $D/4L$.

The complex modulus of elasticity E^* is defined by

$$E^* = E(1+i\eta), \quad (4)$$

where E is the dynamical Young modulus and η is the loss factor of the material.

For the sinusoidal displacement of the foundation

$$\tilde{x}_1(t) = x_1^* e^{i\omega t} = x_1 e^{i(\omega t+\phi_1)}, \quad (5)$$

the resulting displacement of the mass M is equal

$$\tilde{x}_2(t) = x_2^* e^{i(\omega t)} = x_2 e^{i(\omega t+\phi_2)} \quad (6)$$

x_1 and x_2 are amplitudes of the displacements,
$\phi=\phi_2-\phi_1$ — the phase shift between them, respectively.

From the equations (2), (5) and (6) one gets

$$\frac{x_2^*}{x_1^*} = \frac{KE^*-i\omega\beta}{-M\omega^2+KE^*-i\omega\beta} = \frac{1+i\left(\eta-\frac{\beta\omega}{KE}\right)}{\left(1-\frac{M\omega^2}{KE}\right)+i\left(\eta-\frac{\omega\beta}{KE}\right)} \quad (7)$$

and after some calculations one can derive for the transmissibility T and the phase angle ϕ, respectively

$$T = \left|\frac{x_2^*}{x_1^*}\right| = \frac{\left[1+\left(\eta-\frac{\beta}{KE}\omega\right)^2\right]^{1/2}}{\left[\left(1-\frac{M\omega^2}{KE}\right)^2+\left(\eta-\frac{\omega\beta}{KE}\right)^2\right]^{1/2}} \quad (8)$$

and

$$\operatorname{tg}\phi = \frac{-\dfrac{M\omega^2}{KE}\left[\eta - \dfrac{\omega\beta}{KE}\right]}{\left(1-\dfrac{M\omega^2}{KE}\right)+\left(\eta-\dfrac{\omega\beta}{KE}\right)^2} \tag{9}$$

3. Determination of E and β

For the resonance of the simple system in vacuum the angular frequency $\omega = \omega_0$ and it is given by the relation

$$\omega_0^2 = \frac{KE_0}{M} \tag{10}$$

where E_0 is the Young modulus of the sample.

Introducing (10) into the equation (2) and (8) one gets

$$T^2 = \frac{1+\left(\eta - \dfrac{\beta}{KE}\omega_0\right)^2}{\left(\eta - \dfrac{\beta}{KE_0}\omega_0\right)^2} \tag{11}$$

and after the transformation

$$\beta^2 - \frac{2KE_0\eta}{\omega_0}\beta + \frac{\eta^2K^2E_0^2}{\omega_0^2} - \frac{K^2E_0^2}{\omega_0^2(T^2-1)} = D \tag{12}$$

This square equation for β may be easily solved. On the other hand when the measurement of T and ω_0 are performed in vacuum $\beta = 0$ and then η and E may be easy calculated for the resonance frequency ω_0. Next the β value is determined using (12). The evaluated value for the case in the air was obtained as $\beta = 1800$ g/s and this value will be used to the numerical calculations described in the chapter 5.

4. Data for the numerical analyses

The formulae (8) and (9) were used for the numerical calculations of the transmissibility and the phase, respectively of a sample presenting the system of the single degree of freedom.

Different samples of polyurethane material were taken for measurements and for numerical analyses. The samples had the following geometry: the heights were $h = 0.01$ m or 0.02 m and the square cross-section of $A^2 = 10^{-4}$ m².

The preliminary values of the Young modulus E_0 and the loss factor η for the samples were determined in vacuum ($\beta = 0$, according to (10) and (11)) at resonances assuming they have presented single degree of freedom systems.

The samples were numbered from 1 – 4 and all data for them are collected in the Table 1.

Table 1. The data of the samples

Number	Height h [m]	Young's modulus $E \times 10^{-7}$ [N/m²]	Loss factor η	Mass $m \times 10^3$ [kg]	Mass on the sample $M \times 10^3$ [kg]
1	0.019	1	0.35	1.14	22
2	0.02	0.133	0.6	1.05	22
3	0.01	0.154	0.59	0.56	45
4	0.01	0.197	0.78	0.56	22

5. Numerical results

The results of calculations are presented in the following Figures from 2a,b – 5a,b. In every Figure two curves are presented one for the vacuum ($\beta=0$) and the other for the air ($\beta=1800$ g/s) conditions; Figures (a) correspond to characteristics of the transmissibility T and (b) to characteristics of the phase.

In Fig. 2a the results of T for the sample 1 are given. It can be seen that the ratio $\Delta T/T(\beta=0)$ where

$$\Delta T = T(\beta=1800) - T(\beta=0),$$

has its maximum value at the resonance frequency. Also, one can deduce that in the case the loss factor η would be calculated from the formula (11) for $\beta=0$ and in air, then the value of η had been smaller than the real one.

The Fig. 2 b presents the dependence of the phase angle ϕ against frequency for $\beta=0$ and $\beta=1800$ g/s (according to Eq. (9)). The influence of β on ϕ is evident when one have compared the two curves: $\phi(\beta=0)$ and $\phi(\beta=1800)$. The phase difference $\Delta\phi=\phi(\beta=1800)-\phi(\beta=0)$ increases against the frequency to achieve a maximum just before the resonance frequency. After a rapid drop of the phase difference down to the opposite sign in the region of the resonant frequency, the other extremum is achieved and next the gradual (decrease of the curve in the Figure) increase of the absolute value of the phase difference takes place.

It means that the Young modulus E as well as the loss factor η will be influenced by the air conditions when they are determined using Eqs. (8) and (9) valid for vacuum. So, in practice the error due to the neglecting of the air influence increases against the frequency, too.

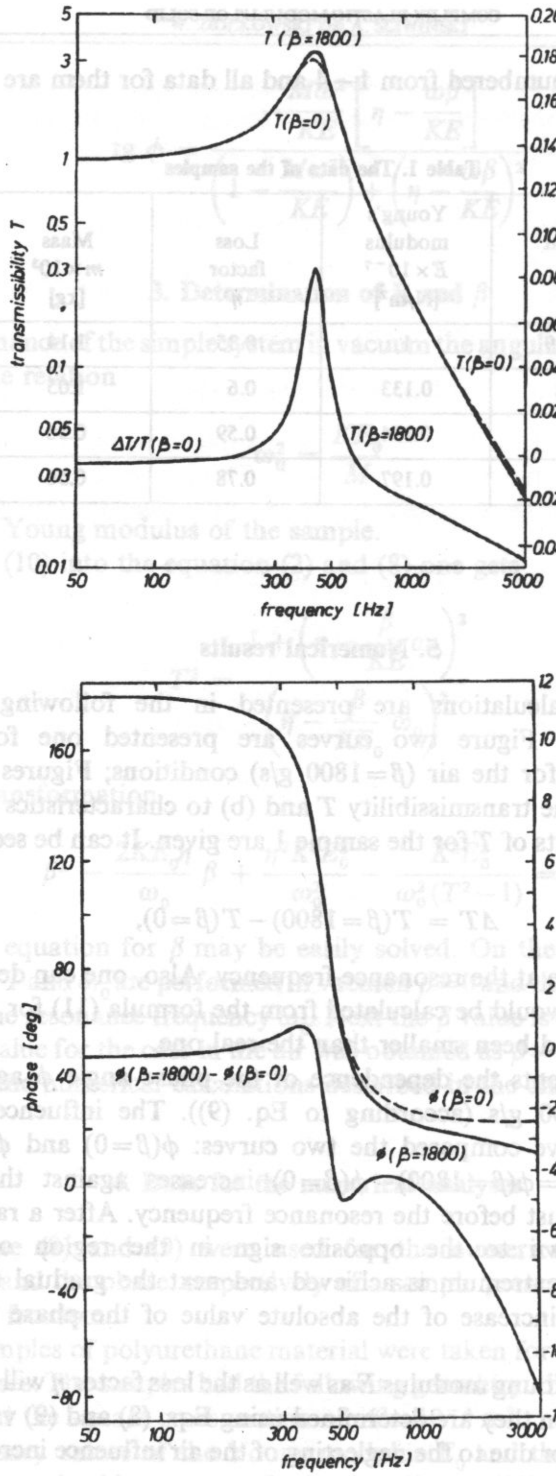

Fig. 2. Frequency dependence of (a) the transmissibility and (b) the phase angle for the sample 1.

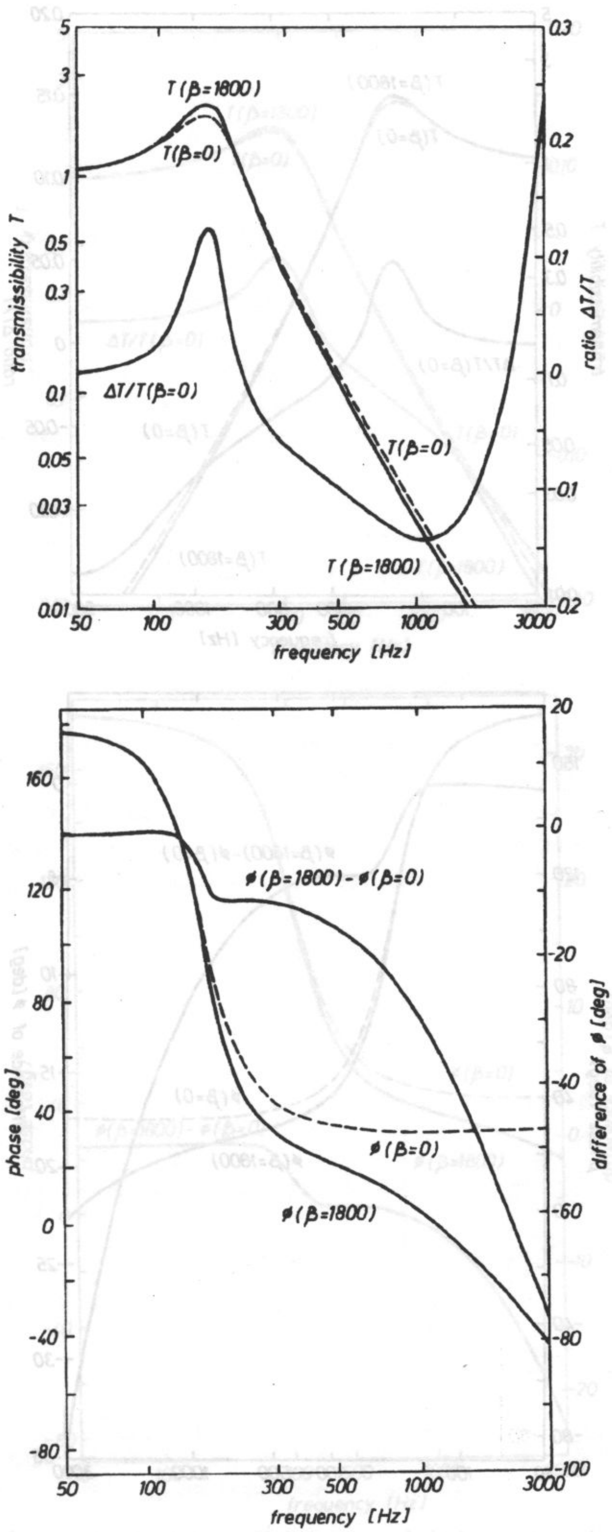

Fig. 3. Frequency dependence of (a) the transmissibility and (b) the phase angle for the sample 2.

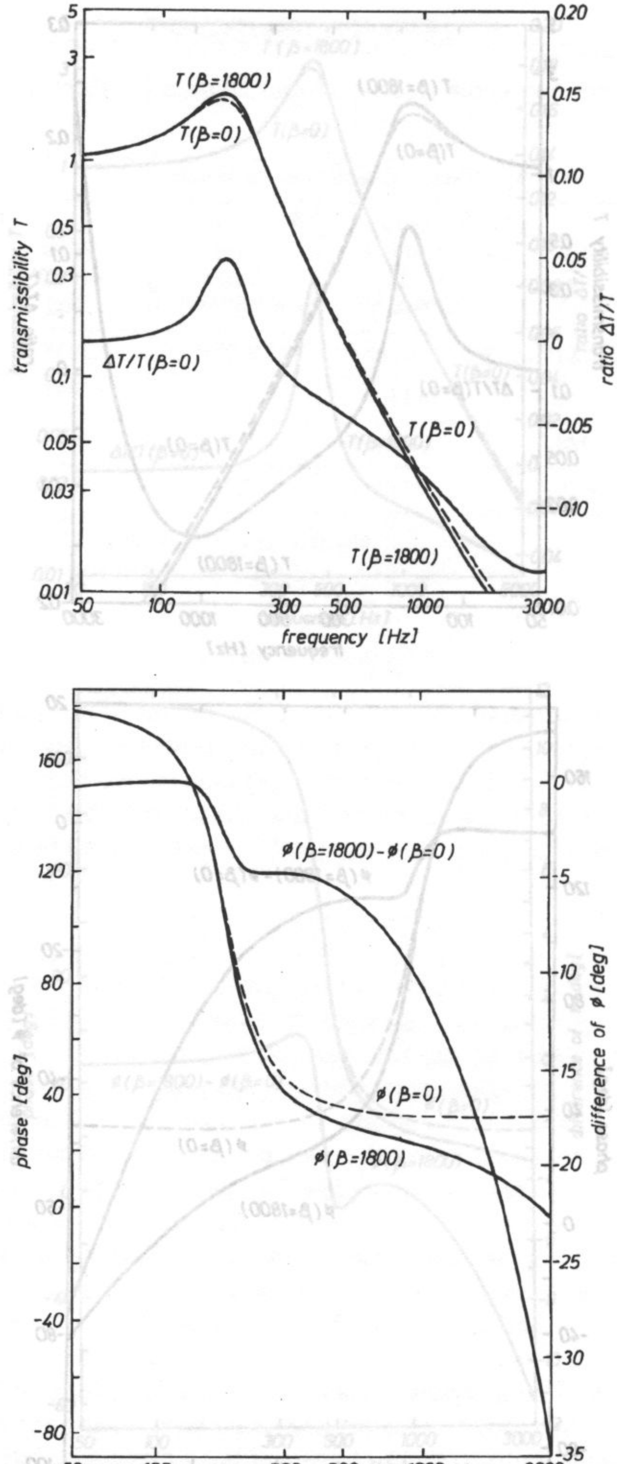

Fig. 4. Frequency dependence of (a) the transmissibility and (b) the phase angle for the sample 3.

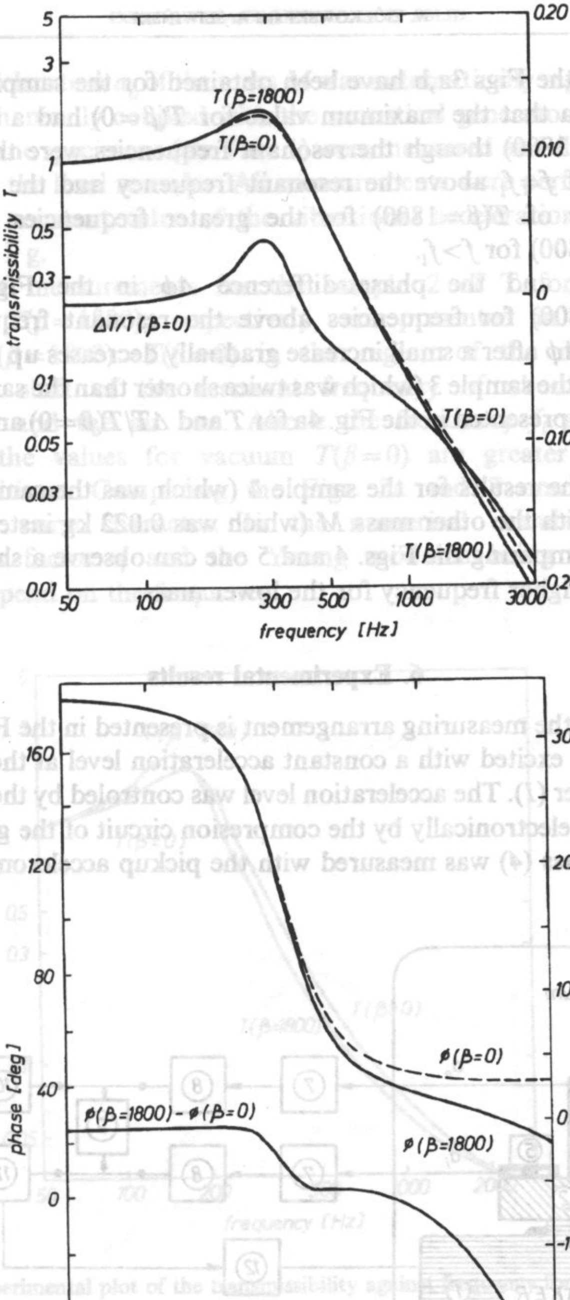

Fig. 5. Frequency dependence of (a) T, $\Delta T/T$ and (b) ϕ, $\Delta\phi$ for the sample 4.

The curves in the Figs 3a,b have been obtained for the sample 2. It is evidently seen in the Fig. 3a that the maximum value for $T(\beta=0)$ had a smaller value than the one for $T(\beta=1800)$ though the resonant frequencies were the same. The both curves intersect at $f=f_i$ above the resonant frequency and the curve for $T(\beta=0)$ lies over the one of $T(\beta=1800)$ for the greater frequencies i.e. the values of $T(\beta=0)>T(\beta=1800)$ for $f>f_i$.

The phase ϕ and the phase difference $\Delta\phi$ in the Fig. 3b, show that $\phi(\beta=0)>\phi(\beta=1800)$ for frequencies above the resonant frequency. Above the resonance region $\Delta\phi$ after a small increase gradually decreases up to the frequency f_i.

The results for the sample 3 (which was twice shorter than the sample 2 but with the some material) are presented in the Fig. 4a for T and $\Delta T/T(\beta=0)$ and in the Fig. 4b for ϕ and $\Delta\phi$.

In the Fig. 5 the results for the sample 4 (which was the same as the sample 3, however loaded with the other mass M (which was 0.022 kg instead of the 0.045 kg — see Tab. 1). Comparing the Figs. 4 and 5 one can observe a shift of the resonant frequency to the higher frequency for the lower mass.

6. Experimental results

The scheme of the measuring arrangement is presented in the Fig. 6. The samples were harmonically excited with a constant acceleration level at the shaking table (*2*) driven by the shaker (*1*). The acceleration level was controled by the accelerometer (*5*) and kept constant electronically by the compresion circuit of the generator (*11*). The response of the mass (*4*) was measured with the pickup accelerometer (*6*).

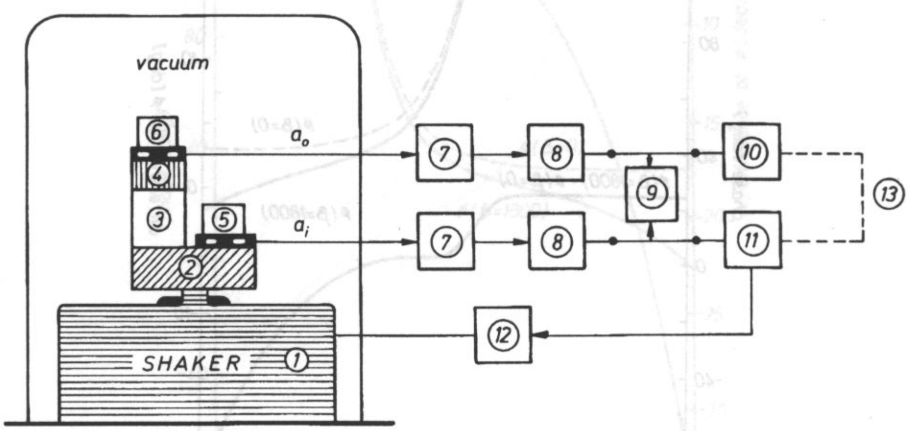

Fig. 6. Measuring arrangement *1*. shaker; *2*. table; *3*. specimen; *4*. loading mass; *5*. control accelerometer; *6*. pickup accelerometer analyzer; *9*. phase meter; *10*. level recorder; *11*. sinusoidal generator; *12*. power amplifier; *13*. mechanical linkage.

The output acceleration a_0 of the mass M was automatically registed with the level recorder (*10*) mechanically coupled with the acoustical generator (*11*).

The polyurethane specimens (see Tab. 1) were measured and the transmissibilities were registed with the level recorder. All measurements were performed at the room temperature. The constant value of the vibrational acceleration amplitude of the shaking table $a_i=1$ g.

The results of measurements for the sample 2 of T for vacuum $T(\beta=0)$ and in the air $T(\beta=1800)$, respectively, are presented in the Fig. 7. One can see, that $T(\beta=1800)>T(\beta=0)$ in the region of the resonant frequency f_r. There was a shift of the resonant frequency of about 12 Hz between the vacuum case and the air one. Above the frequency f_i at which the both curves intersect the values for vacuum $T(\beta=0)$ are greater than for the air $T(\beta=1800)$ conditions. Comparing the Figs. 3a and 7 one can say that the curves have the same character. In the numerical calculations, there were assumed the loss factor η and the Young modulus being constant, however the real values depend on the frequency.

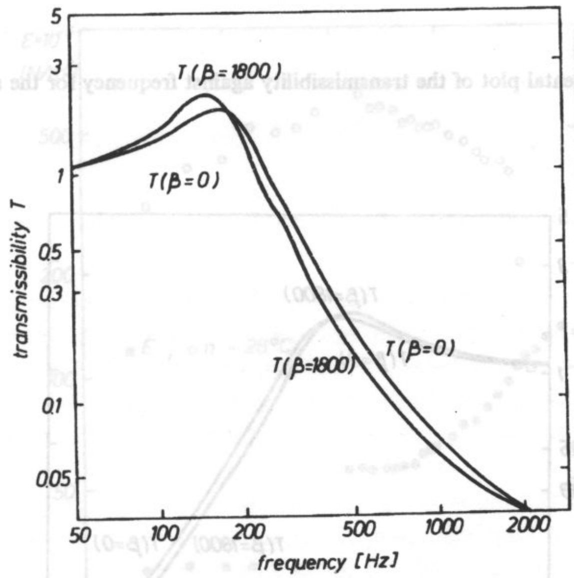

Fig. 7. Experimental plot of the transmissibility against frequency for the sample 2.

Also, in the experiment some influence of additional damping of the glue joining the specimen and the testing device was observed, however it was neglected in the theoretical considerations.

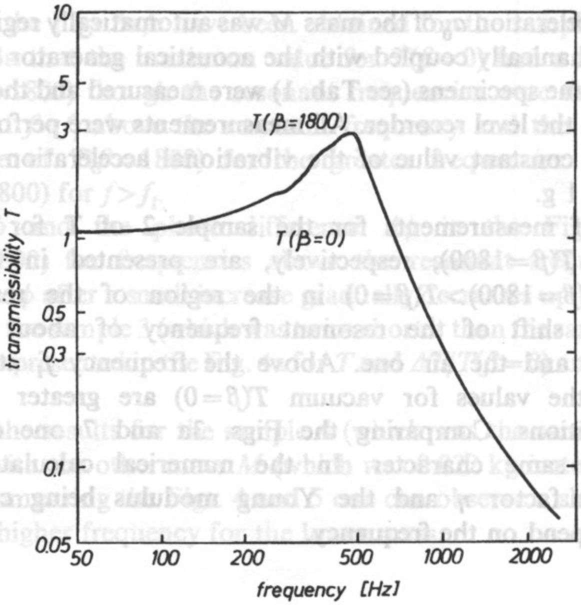

Fig. 8. Experimental plot of the transmissibility against frequency for the sample 1.

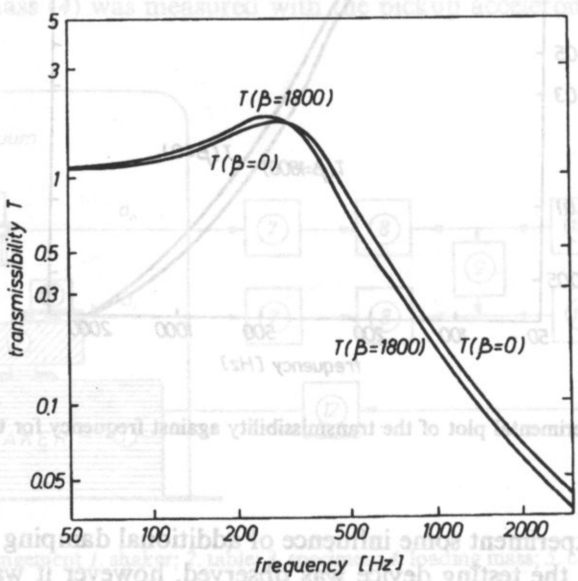

Fig. 9. Experimental plot of the transmissibility against frequency for the sample 4.

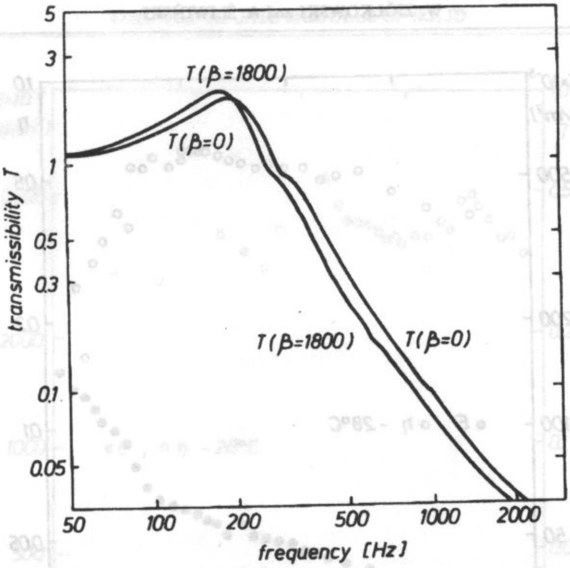

Fig. 10. Experimental plot of the transmissibility against frequency for the sample 3.

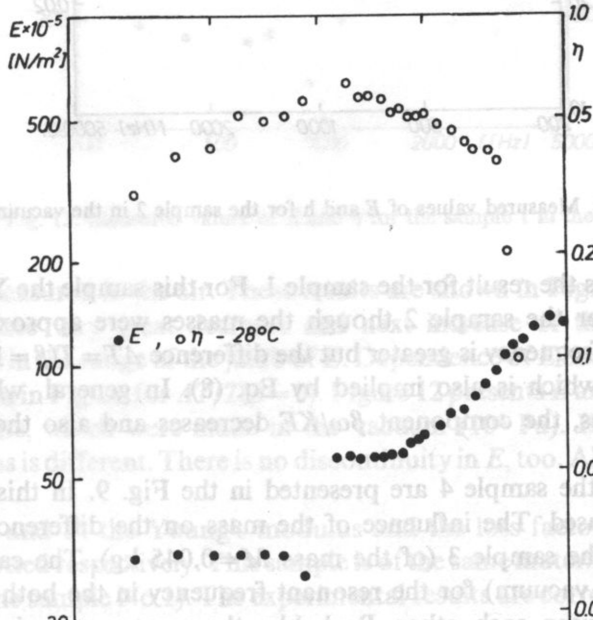

Fig. 11. Dynamic Young's modulus and loss factor of the soft polyurethane from (the same material as the sample 2) plotted against frequency.

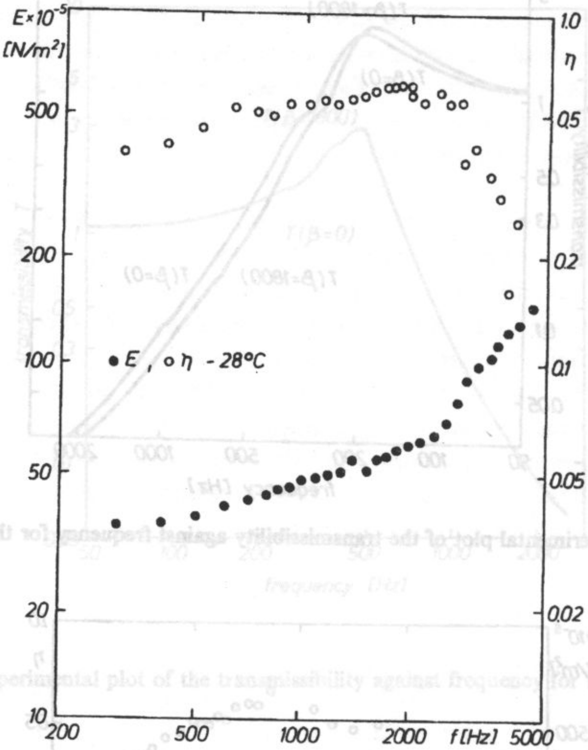

Fig. 12. Measured values of E and h for the sample 2 in the vacuum.

The Fig. 8 shows the result for the sample 1. For this sample the Young modulus was greater than for the sample 2 though the masses were approximatelly equal. Now, the resonant frequency is greater but the difference $\Delta T = T(\beta=1800) - T(\beta=0)$ is practically zero which is also implied by Eq. (8). In general, when the Young modulus E increases, the component $\beta\omega/KE$ decreases and also the difference ΔT decreases.

The results for the sample 4 are presented in the Fig. 9. In this case the mass $M=0.022$ kg was used. The influence of the mass on the difference ΔT is shown in the Fig. 10 for the sample 3 (of the mass $M=0.045$ kg). The calculated values for η and E (in the vacuum) for the resonant frequency in the both samples 3 and 4 are different between each other. Probably, the greater mass introduced some changes in the structure of the sample. Typical experimental results E and η for the analysed samples obtained by the method are described here. It was used the sample of the soft polyurethane foam; the same material what the specimens 1, 3 and 4 was made.

The dynamic Young's modulus E and the loss factor η were calculated from the system of the equations (8) and (9). In calculations β were neglected, whereas the values

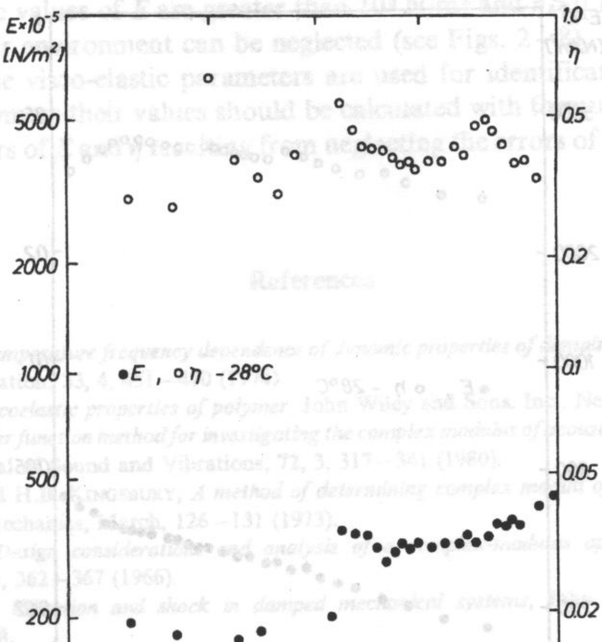

Fig. 13. Measured values of E and η for the sample 1 in the air.

T and ϕ were measured in the air. These results are shown in Fig. 11 for temperature $T_1 = 28°C$. We see very great decrease and next increase of the value E and the maximum value in the range of the jumb of E. Dependence of E has the same character as have been seen in Fig. 2a for $\Delta T/T(\beta = 0)$. Figure 12 presents E and η values obtained by measurements, which were made in the vacuum (10^2 Pa). Here the plot of the Youngs modulus is different. There is no discontinuity in E, too. Also, the values E and η are smaller.

At Figs. 13 and 14 the Young's modulus and the loss factor in the air and the vacuum, are plotted respectively. This sample is of the same material as used to the plot of T at Fig. 8 (the sample No 1). The experimental results are correct up to about 2000 Hz; this agrees with a condition that a longitudinal wave $\lambda_L = \sqrt{\frac{E}{\rho}} \Big/ f$ is less than $h/8$. It is equivalent that errors in calculations of E and η are less than 10%. Above this longitudinal wave a formula for a transmissibility of a rod [6] must be used to calculate Young's modulus and loss factor.

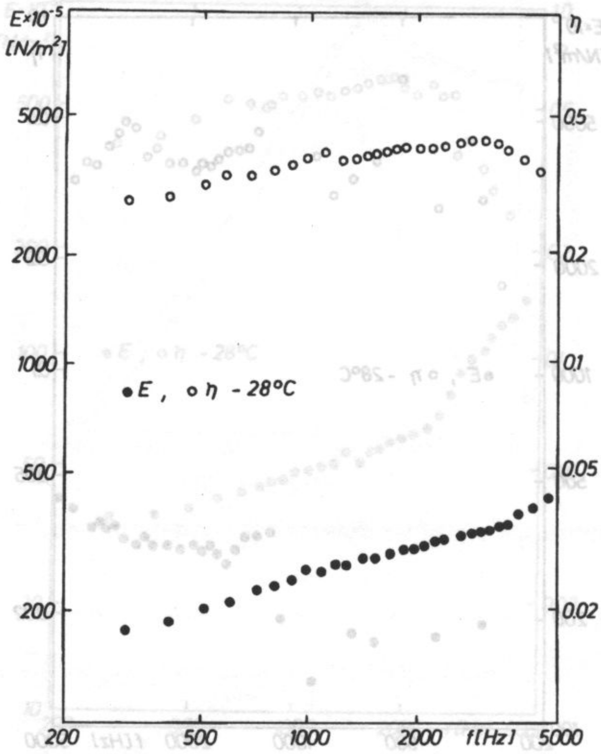

Fig. 14. Frequency dependence of E and h for the vacuum for the sample 1.

7. Conclusions

Theoretical considerations, numerical analysis and some experimental examination leads to the following conclusions:

1. The equations (8) and (9) show that for the determined values of η, E and ω characterizing the visco-elastic properties, the influence of the air damping can be essential or nonessential contribution to the calculation of complex transmissiblity; the situation depends on those values.

2. The essential influence of the air damping on the measuring of the complex transmissibility is observed for samples of small densities, $\rho < 6\ 10^2\ kg/m^3$, Young modulus $10^6\ N/m^2$ and loss factor equal 0.6. It is seen from the Figs 3a–5a presenting numerical curves as well as from experimental transmittance dependences for normal conditions and for vacuum (10^2 Pa) that the measured values of Young modulus and loss factor are different for this different conditions (see Figs. 11–14).

3. In case the values of E are greater than 10^8 N/m² and $\eta > 0.1$ the contribution from losses of air environment can be neglected (see Figs. 2–8).

4. In case the visco-elastic parameters are used for identification of relaxation processes in polymers their values should be calculated with formulae (8) and (9). In practice the errors of E and η resulting from neglecting the errors of air friction should be determined.

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