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DESIGNING PROBLEMS IN COMPUTERIZED ACOUSTIC ANALYZERS

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An attempt to present considerable development in the domain of computerized acoustic devices is made. The synthesis refers mostly to computer-based acoustic analyzers leading in that domain because of their versatility. The mathematical basis as well as the main test functions are reviewed. Hardware and software assumptions of the most universal computerized equipment are shown. The example of an FFT analyzer realized on the basis of the LabVIEW software package for the Macintosh II computer is presented. Conclusions and index of bibliography are included.

1. Introduction

On the basis of the latest advances in computer technology, it is possible to combine all necessary test functions in a single multi-purpose instrument and at the same time to improve operation and measurement procedures. Such a system can be easy-adaptive to any kind of acoustical measurement.

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universal acoustic analyzer is the selection of its test functions. Equally important is the proper mathematical description of the analyzed physical phenomena. Mathematical formulas are to be represented by the algorithms used as the basis of procedures written in programming languages or machine codes. However, those procedures are not equally effective. This is of great importance when both the ease of programming and the speed of processing are concerned.

A computerized measuring device provides a powerful technique for collecting, analyzing and presenting data in an organized and systematic fashion. Nevertheless, a considerable amount of expertise in electronics and computers is necessary to configure a data acquisition system. That problem is particularly interesting because of the variety of configurations in existing measuring systems.

It is possible to discern a few types of computerized acoustic measuring devices. The first group comprises all multi-function analyzers dedicated to the domain of a single application: e.g., architectural acoustics, ultrasonics, studio technology, mechanical vibrations etc. The next group includes systems in which a computer is applied in the control of the measurement process and, additionally, makes possible communication with the user.

The introduction of personal computers to the fast processing of acoustic signals has affected in a significant way the topology of the typical measuring laboratory. A computer provided with data acquisition cards and supporting software becomes a powerful tool to perform even a very sophisticated analysis.

The mentioned solution can be considered to be the most advanced in the domain of digital measuring systems and, at the same time, economically recommended.

The main topics of the following discussion is the dependence of the system capabilities on its architecture and on the employed programming method.

2. Analysis of quantized acoustical signals

Acoustic measurements mostly require time and frequency analyses of signals. Moreover, the simultaneous analysis of two functions e.g. excitation and response signals, justifies the design of analyzers as dual-channel.

Analytical signals in the domain of time can be displayed in each channel in terms of: their real and imaginary part, their magnitude and phase versus time, their NYQUIST or NICHOLS plots.

The analytical signal corresponding to the real-valued time record of input can be represented by

$$x_z(n) = x(n) + j\tilde{x}(n) \tag{1}$$

where n — number of samples of signal record.

As the following time records $x(n)$ are to be averaged, the analytical signal can be expressed by

$$\bar{x}_z(n) = \bar{x}(n) + j\tilde{\bar{x}}(n) \tag{2}$$

$$\tilde{\bar{x}}_z(n) = \mathscr{H}[\bar{x}(n)] \tag{3}$$

$$\bar{x}_z(n) = \mathfrak{F}^{-1}[G(k)] \tag{4}$$

where k — frequency index, $\mathscr{H}$ — discrete Hilbert transform, $G(k)$ — instantaneous spectrum, $\mathfrak{F}^{-1}$ — inverse fast Fourier transform.

On the basis of Eq. (4) — the autospectrum function $\bar{G}_{11}(k)$ can be calculated as follows:

$$\bar{G}_{11}(k) = \bar{G}_1^*(k)\bar{G}_1(k) = |\bar{G}_1(k)|^2 \tag{5}$$

where $G_1^*(k)$ — complex conjugated spectrum.

The $|\bar{G}_1(k)|^2$ can be scaled and furtherly interpreted as: rms, power, power spectral density or energy spectral density of the signal $x(n)$. Those parameters provide an important representation of acoustical signals.

Calculation of the response of acoustical systems to different excitations is permitted on account of designing an analyzer as dual-channel.

Therefore the cross spectrum function of both channels should be calculated:

$$\bar{G}_{12}(k) = G_1^*(k)G_2(k) = G_2^*(k)G_1(k). \tag{6}$$

Using the latest equation various types of frequency responses could be calculated from

$$H_1(k) = \frac{\bar{G}_{12}(k)}{\bar{G}_{11}(k)}. \tag{7}$$

The importance of Eq. (7) results from its physical interpretation. The complex ration of the averaged cross spectrum $\bar{G}_{12}(k)$ and the averaged autospectrum $\bar{G}_{11}(k)$ reflect the gain and phase lag introduced by a linear system in response to the $x_1(n)$ excitation.

The frequency response can be expressed as magnitude, phase or plots in Nyquist or Nichols coordinates.

The convolution of the signals in both channels, represented in the frequency domain as a multiplication of spectra, can be used to determine the coherence between those signals.

As an indication of the quality of the measurement, one may use following formula:

$$C_{12}^2(k) = \frac{|\bar{G}_{12}(k)|^2}{\bar{G}_{11}(k)\bar{G}_{22}(k)}. \tag{8}$$

That factor expresses the level of linear dependence between the signals in both channels of the analyzer related by a linear frequency response of the measuring object to the excitation $x_1(n)$.

The next parameter, i.e., signal to noise ratio, determines one of the most important features of an audio channel. Moreover, like a coherence factor that parameter related by a linear frequency response function to the signal $x_1(n)$ may be used to indicate the quality of the performed measurement.

The signal to noise ration is defined by the expression

$$\frac{S}{N}(k) = \frac{C_{12}^2(k)}{1 - C_{12}^2(k)} = \frac{\dfrac{|\bar{G}_{12}(k)|^2}{\bar{G}_{11}(k)\bar{G}_{22}(k)}}{1 - \dfrac{|\bar{G}_{12}(k)|^2}{\bar{G}_{11}(k)\bar{G}_{22}(k)}} = \frac{|\bar{G}_{12}(k)|^2}{\bar{G}_{11}(k)\bar{G}_{22}(k) - |\bar{G}_{12}(k)|^2} \tag{9}$$

and may be used to describe the signal dynamics.

As it results from the modern auditory theory, the perception process cannot be properly considered without taking into account the binaural cross-correlation phenomena. Hence, providing the measuring system with the correlation analytical functions becomes substantial.

The basic function in the domain of correlation analysis is the autocorrelation function defined in the spectrum averaging mode as follows:

$$R_{11}(n) = \mathfrak{F}^{-1}[W(k)\bar{G}_{11}(k)] \tag{10}$$

where $W(k)$ is the weighting function of the autospectrum function.

In order to make possible the comparison of various signals, the normalization of autocorelation values is desired. That operation is defined by

$$\varrho_{11}(n) = \frac{R_{11}(n)}{R_{11}(0)} \tag{11}$$

Autocorrelation can be presented in terms of real and imaginary parts, magnitude and phase or pole coordinates.

Similarly the cross-correlation function is defined by

$$R_{12}(n) = \mathfrak{F}^{-1}[W(k)\bar{G}_{12}(k)]. \tag{12}$$

In that case the normalization leads to the definition of the cross-correlation coefficient as follows:

$$\varrho_{12}(n) = \frac{R_{12}(n)}{\sqrt{R_{11}(0)R_{22}(0)}} \tag{13}$$

where

$$R_{11}(0) = \frac{1}{N}\sum_{k=0}^{N-1} W(k)\bar{G}_{11}(k) \tag{14}$$

$$R_{22}(0) = \frac{1}{N}\sum_{k=0}^{N-1} W(k)\bar{G}_{22}(k) \tag{15}$$

According to the principles of the circuit and signal theory the impulse response can be used as the basic parameter of the transmission channel. Hence, its presence in the function menu of an analyzer cannot be neglected.

In the spectrum averaging mode, the impulse response can be defined by the equation

$$h(n) = \mathfrak{F}^{-1}[W(k)H_1(k)] \tag{16}$$

The impulse response of a linear system can be determined by exciting the system with white noise and cross-correlating the input and output. Although the most direct approach is to apply an impulsive excitation such as electronic spark gaps, pistol shots or exploding balloons, it is difficult, nevertheless, to provide that kind of signal with sufficient energy, repeatable in terms of its amplitude and directional characteristics.

In order to overcome these problems and to minimize the amount of computation required by the cross-correlation operation, the system can be excited by a binary maximum-length sequence, and the cross-correlation performed using the fast Hadamard transform [6], [7], [8], [23].

Several of the references explain how to generate maximum-length sequences based upon a primitive polynomial [9], [23].

The mathematical operations given above enable calculations of the majority of acoustical parameters, They permit, among others, to calculate sound power, sound absorption and insulation, reverberation time according to various standards [30], still spectrum manipulation such as: arithmetic operations, integration, weighting, differentiation, spectrum weighting etc.

Having calculated the discrete spectrum of an acoustic signal it is possible to display the measurement results in the form of octave or third-octave analysis by averaging spectral energy in the corresponding frequency bands.

In the domain of electroacoustic equipment technology, many additional test functions are required for quality test applications. These parameters which are presented in Fig. 1 can be calculated on the basis of the discrete FFT being the standard operation during signal processing in a digital analyzer.

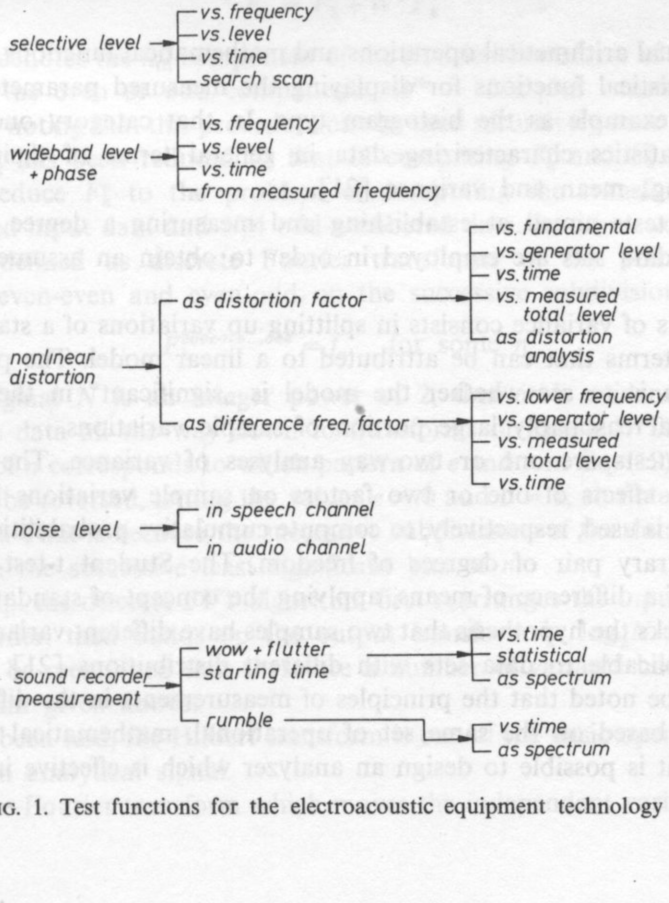

FIG. 1. Test functions for the electroacoustic equipment technology

As it is shown in Fig. 1, the amount of test functions is considerable, so the mathematical description will be restricted to the most important ones. The remaining parameters are defined in the bibliography [16], [22], [30].

The determination of the nonlinear distortion factor according to the definition given in Eq. (17) is a standard procedure when testing an audio channel

$$h_{0/0} = \frac{\sqrt{h_1^2 + h_2^2 + \ldots + h_n^2}}{n} \tag{17}$$

where $h_1 \ldots h_n$ are the subsequent harmonics of the signal extracted from the discrete spectrum.

In that kind of analysis the difference frequency factor is usually carried out. The IEC total difference-frequency distortion TDFD test is a recently proposed method for detecting and quantifying a broad range of nonlinear distortions in audio equipment [24]. The test is devised to be sensitive to both asymmetrical and symmetrical nonlinearities and to both static and dynamic distortions [24]. The test-signal and primary distortion-product spectra can be shown as results of the distortion analysis in a numerical form or can be plotted versus level, frequency, or time.

Besides usual arithmetical operations and mathematical functions, it is necessary to set up statistical functions for displaying the measured parameter in terms of statistics, for example as the histogram type. In that category one may include descriptive statistics characterizing data in general terms of their moments of distribution e.g., mean and variance [21].

Statistical tests aimed at establishing and measuring a degree of correlation between two data sets are employed in order to obtain an assumed accuracy of calculations.

An analysis of variance consists in splitting up variations of a statistical sample into a set of terms that can be attributed to a linear model. The purpose of this decomposition is to see whether the model is „significant" in the sense that it accounts for a reasonably large portion of sample variations.

ANOWA tests are one or two-way analyses of variance. They are used to determine the effects of one or two factors on sample variations [21].

The F-test is used, respectively, to compute cumulative probabilities and fractiles with an arbitrary pair of degrees of freedom. The Student t-test measures the significance of a difference of means, applying the concept of standard error while the F-test checks the hypothesis that two samples have different variances. There are also tests applicable to data sets with different distributions [21].

It should be noted that the principles of measurements in the different areas of acoustics are based on the same set of operational, mathematical and statistical functions, so it is possible to design an analyzer which is effective in any acoustic domain.

3. Algorithmic representation of the main analytical procedures

As it results from the quoted mathematical formulas, processing of the measured signals demands, first of all, the following operations in both frequency and time domains: FFT, FFT^{-1}, Hilbert and Hadamard transforms.

If the discrete FFT is performed using its definition, then the execution time requires n^2 multiplications, where n is the number of signal samples. In order to minimize the amount of computation, one can use the fast Fourier transform based upon the algorithms of Cooley and Tukey and, further, of Winograd [21]. The recently proposed methods eliminate the number of operations and as a result they speed up the processing hence addressing of data becomes more complicated. In fact, the discrete Fourier transform can be computed in $n\log_2 n$ operations.

The fast Fourier transform can be described in terms of a matrix multiplication. Its routine is based on the flow graph known also as butterfly graph [10, 20, 21].

Danielson and Lanczos showed that a discrete Fourier transform of length N can be rewritten as the sum of two discrete Fourier transforms, each of the length $N/2$ [21]. One of them is formed from the even-numbered points of the original N, the other one from the odd-numbered points:

$$F_k = F_k^e + W^k F_k^o \tag{18}$$

where F_k^e, F_k^o denotes the k_{th} component of the Fourier Transform of the length $N/2$ formed from the even or odd components, W^k — complex constant.

It is worth noting that this procedure can be used recursively once the problem of computing F_k has been reduced to that of cumputing F_k^e and F_k^o. Similarly it is possible to reduce F_k^e to the problem of computing the transform of its $N/4$ even-numbered input data and $N/4$ odd-numbered data. In other words, FK^{ee} and F_k^{oe} can be defined as discrete Fourier transforms of the points which are, respectively, even-even and even-odd on the successive subdivisions of the data

$$F_k^{eoeeoeo\ldots oee} = f_n \quad \text{for some } n \tag{19}$$

When original N is an integer power of 2, then it is evident that one can subdivide the data all the way down to transforms of length 1. Then, to identify which value of n corresponds to which pattern of e and o in Eq. (19), the pattern of e and o is to be reversed, letting the values $e = 0$ and $o = 1$, so obtaining in binary the value of n. That is because the successive subdivisions of the data into even and odd tests are the successive least significant bits of n.

To sum up: the discrete FFT algorithm first rearranges the input elements into bit-reverse order, then builds up the output transform in $\log_2 N$ iterations.

It should be mentioned that there are a number of variants built on the basic FFT algorithm given above.

As it has been said, the Hilbert transform is one of the basic operations required to process an analytical signal.

Unlike the Fourier transform which moves the independent variable of a signal

$x(t)$ from the time domain to the frequency domain, the Hilbert transform leaves the signal $x(t)$ in the same domain. The Hilbert transform $\tilde{x}(t)$ of a real-valued time signal $x(t)$ is another real-valued time signal, and the Hilbert transform of a complex-valued frequency function $X(f)$ is another complex-valued frequency function.

The Hilbert transform can be defined as a convolution integral' phase shift system or as the imaginary part of an analytic signal. In practice such an integral is to be replaced by finite summations [5].

Fast Madamard transform can be considered after FFT, as the most important transformation. When considering an acoustic measurement system excited by a pseudorandom noise, it is very efficient to use that transform in order to measure its impulse response [6], [8].

Like the discrete Fourier transform, the Hadamard transform can be described in terms of matrix multiplication [8], [9], [23]. The matrix that transforms the input vector is known as the Hadamard matrix H_n where n gives the number of rows or columns. The elements of the Hadamard matrix are all $\pm$ 1, and the matrix must satisfy the relation

$$H_n H_n^T = nI_n \tag{20}$$

where I_n – unit matrix

The algorithm applies only to the specific class of Hadamard matrices known as the Sylvester-type. The Hadamard matrix is defined recursively by [8]

$$H_1 = [1]$$

$$H_{2i} = \begin{bmatrix} \mathrm{H}_i & \mathrm{H}_i \\ \mathrm{H}_i & -\mathrm{H}_i \end{bmatrix}. \tag{21}$$

Only orders 2^k, where k is a nonnegative integer, exist.

The basic butterfly element for the fast Hadamard transform is shown in Fig. 2.

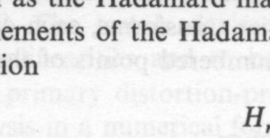

FIG. 2. Basic butterfly element.

It is evident that the flow graph is identical to the flow diagram for the FFT except that the twiddle factors are all unity, reflecting the fact that no multiplications are required. Then it is possible to adapt any Fourier transform routine to execute the fast Hadamard transform [8].

The fast Hadamard transform requires only about 2.5 n$\log_2 n$ operations which in this case are additions. Hence, there is no need for multiplications that represent a reduction in the execution time [8].

The speed of processing depends on the type of operation being performed (matrix multiplications or additions) and the type of computer beind used. The time required to execute FFT in one of the most powerful realization (see Paragraph 5), where the FFT function is defined by the following summations:

$$F(i) = \sum_{k=0}^{N-1} \text{array! } (k) * e^{-j2\pi ki/N}, \tag{22}$$

is about 7 s for the power 8, while VFFT (very fast Fourier transform) is approximately 100 times faster than FFT. The quoted parameters concern the McADIDS system on the Macintosh Plus computer [15].

4. System configuration problems

Contemporary analyzers develop, in qeneral according to the configuration of applied microprocessor systems.

The traditionally designed analyzer provides a detached measuring device which is assigned to collect and process measured values and to display results. The main insufficiency of such a device usually results from the lack of its versatility. Thus, in the case of more complex tests a certain number of similar devices has to be used simultaneously. As the tests results should be stored, printed or plotted, too many devices and too many interconnections between them are needed. This gives rise to a nonoptimal structure of the instrumentation system.

In the domain of acoustics, compactness and portability are of main importance. This issues from the fact that there is a need to transport those devices for outdoor tests in architectural acoustics and from the usual lack of space for storing them in sound recording studios. The principles of processing of measured signals considered in the previous section prove that the great majority of acoustic tests are based on a restricted number of processing routines. Thus, it is recommended to design acoustical analyzers as really universal ones.

Contemporary computer systems create the possibility of fulfilling those requirements but the optimal lay-out of an instrumentation system has not been defined so far. Thus that problem as being current and vital should be thoroughly examined.

Historically, data acquisition and control tasks were implemented either with large mainframe or with minicomputer system. Typically, there were powerful 16-bit machines that ran in time-sharing or multitasking modes. The development of the Personal Computer has enabled to take advantage of the flexibility and efficiency of computerized data acquisition and control. A universal complaint, however, is that a considerable amount of expertise in electronics and computers is necessary to configure a data acquisition system. Nevertheless, owing to a significant degree of standarization among PC and data acquisition and control systems, a large family of hardware and software tools has evolved. The general block diagrams of the computer data acquisition and control system are shown in Figs. 3 and 4.

An important part of any data acquisition system is its host computer. There are

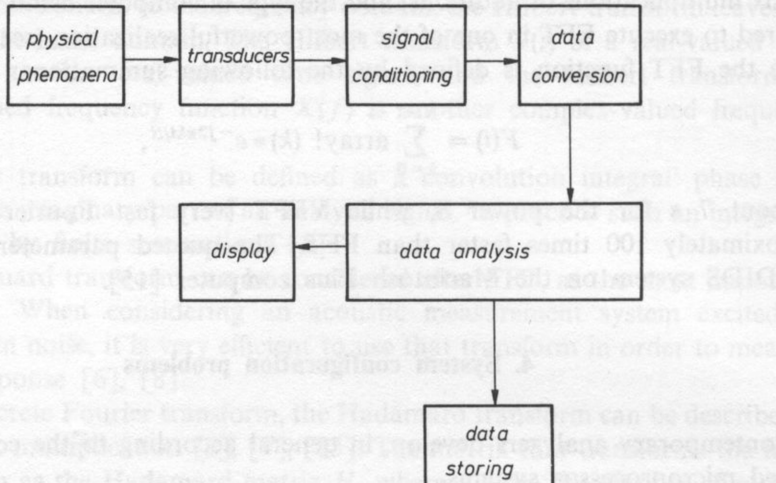

FIG. 3. Data acquisition and control flow diagram

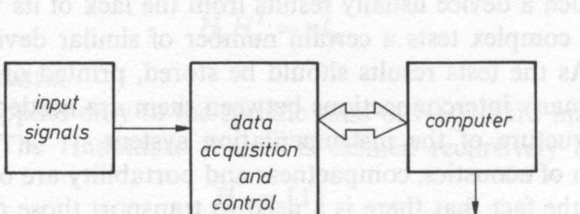

FIG. 4. Computerized data acquisition and control system

two possible ways to interface that system with a computer: connection via a standard interface such as RS-232 (RS-442) or IEEE-488 (see Fig. 5) or direct connection to PC bus (Fig. 6).

The first solution is recommended when the data acquisition system should have the possibility of being interfaced to any type of computer, could be placed remotely from it or when the configured system size is extra complex.

The GPIB (General Purpose Interface Bus) recognized as the IEEE-488 standard has been proven to be one of the most successful and widely accepted interfacing schemes [3]. First of all, the GPIB reduces the hardware requirements of the computer to one interface for each instrument so the main assumption concerning the preferable universal lay-out of the instrumentation system can be fulfilled.

Choosing an interfacing scheme remains one of the most troublesome tasks that engineers, researchers and system designers face, while the importance of the choice

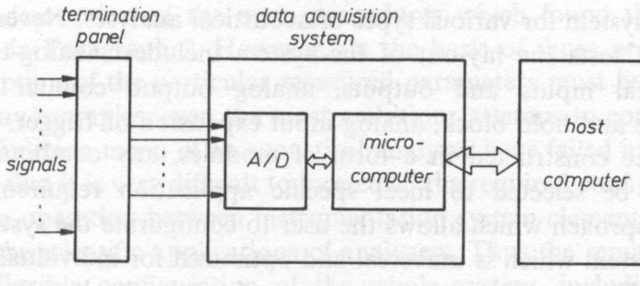

FIG. 5. Block diagram of an external bus data acquisition system

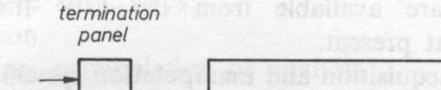

FIG. 6. Internal PC bus system

grows even greater as computers are applied increasingly to complex acoustical phenomena. An inflexible, slow or otherwise inappropriate interface can limit even the most powerful computer-based systems. Interfaces dating earlier than GPIB needed highly specialized hardware and were usually developed for a single purpose. Often those interfaces could not be adapted to even slightly different applications. In the end when the lay-out of universal acoustic instrumentation system is discussed only both previously standard interfaces should be taken into consideration. This follows from the prevalence of RS-232 and advantages of GPIB IEEE-488. According to that principle, several professional instrumentation systems have been designed [27], [29].

The main advantages of a direct connection of the data acquisition system to the PC bus are a data speed of high transmission and instrumentation system compactness. The last mentioned feature is particularly obvious, when the system is composed of computer cards which can be mounted directly into the PC mainframe. The PC expanded in that way provides a powerful platform for collecting, analyzing and presenting data in an organized and systematic fashion. The expansion card can use its own signal processor responsible for the acquisition and fast processing of data [25].

However, the key to the full universality of the discussed system in the domain of acoustics is its modularity. It results from the idea of re-configuring slightly the instrumentation system for various types of acoustical analyses. Nevertheless, in the great majority of tests the lay-out of the system includes: analog-to-digital data acquisition, digital inputs and outputs, analog output, counter (timer) pulse generator, sample and hold block, analog input expansion or trigger. Providing the quoted blocks are constructed in a form of modules, any combination of those instruments can be selected to meet specific application requirements. This is a considerable approach which allows the user to configurate the system by himself and leads to a system which is universal and optimized for individual requirements at the same time. That opportunity maximizes performance and minimizes cost since only those modules are to be installed, which are required for a satisfying performance of the entire range of the involved measurement tasks. The modular instrumentation systems are available from the GW Instruments and the Burr-Brown Corporation at present.

The MacADIOS data acquisition and manipulation system [18] elaborated by the GW Instruments (Massachusetts Institute of Technology) turns the 32-bit Macintosh II computer into a powerful analyzer with capabilities surpassing even the most specialized acoustical measurement systems. The MacADIOS II carrier card plugged into one of the Macintosh expansion slots provides analog input and output channels, digital input and output channels and counter/timer channels. There are also available additional modules, of which up to three can be installed on the carrier to perform such tasks as high-speed sampling and multi-channel input and output.

The IBM compatible personal computer can also co-operate with the similar kind of data acquisition and control system: Burr-Brown's PCI-20000 [27]. The basic configuration of that system is also modular.

As it can be seen from the above review of possibilities given by modern computer technology systems which consist of individual instruments having a front panel with its own combination of indicators, knobs and switches are going to be in defensive. Therefore producers of similar traditional instruments are compelled to revise their approach to the designing of acoustic analyzers.

As to the example of 2032, 2034, 2123 and 2133 Brüel and Kjaer analyzers [11], [17], [26], it becomes obvious that the difference between a computer and an analyzer decreases dramatically. The last mentioned model includes a screen monitor, keyboard and diskette drive. Nevertheless, the versatility of a properly extended standard PC seems to be much better than that of such an analyzer. As a result further development of acoustical analyzers based on PC may be expected.

It is obvious that the capabilities of any computer analyzer are strictly related to their software performance. These problems will be discussed below.

5. Software organization

The most important problem which is to be solved when designing a multi-purpose acoustic analyzer is the proper selection of its test functions. The mathematical description of the main procedures which found the test functions was presented in Paragraph 2. However, on the basis of those procedures a more detailed description of the particular measured parameters must be defined. As it is proven by many examples, even the most ambitious attemps to construct the really universal test function menu of an acoustical analyzer have failed in practice. This is due to the fact that it is very difficult to foresee all the required configurations, ranges and ways of co-operation between instrumentation system elements. This concerns, in particular, the scientific applications of analyzers. Thus the remaining solution is to choose a flexible configuration of the whole system, including the software organization.

The best strategy for high performance is to construct independent procedures for the following three basic steps:
– data acquisition
– data processing, manipulation or analysis
– presenting the data in tabular, graphical or report format.

Each of the acquisition procedures should support a corresponding instrument module which is available in the system. Thus user-callable routines should allow users to specify parameters such as gain, range etc., and to define input/output channels. Those routines are expected to be executable as high-level commands in high-level language such as Basic INPUT statement, and then receive a data string from the acquisition unit.

To enable the processing, manipulation and analysis of data, the instrumentation system based on a personal computer should be designed in such a way that the users have the opportunity to write specialized programs for data acquisition, storage display, logging and control in high-level languages. That possibility demands the use of a special supporting software which links high-level languages to data acquisition subroutines.

Three classes of that software can be discerned: program development tools, function subroutine librairies and complete application packages.

Program development tools and function libraries are used to facilitate writing the unique software. They usually include drivers that provide the interface to the hardware acquisition system. Those software interfaces are intended to establish communication with a data acquisition system using Basic, C or Turbo-Pascal commands. The specially designed data analysis languages such as ASYST [3] or MacADIOS [19] are based on function libraries which include basic procedures used in data acquisition and processing, e.g., data reading, signal filtering, FFT etc.

Complete application software packages are designed to get the system working immediately, usually with no programming required. However, those routines can also offer users facilities to modify the software to meet their own needs. Nevertheless, they are less flexible than the other classes of software.

The most advanced solutions, making possible to create optional instrumentation systems by the user, not necessarily introduced to the programming of a computer, are offered by the software developed by National Instruments Inc. [15],

[27]. Each of these products are intended to run on one of the two main leading PC's: IBM and Macintosh.

The LABTECH is the vast program package which enables the reduction of data acquisition to menu-driven choices. Programming options give users automation and customization of advanced analysis. Digital input data can be recorded and displayed in real-time. Since LABTECH is menu-driven and extremely easy to learn and use, it requires very little computer skills on the part of the operator.

LabVIEW, which is designed for Macintosh computers, replaces conventional programming techniques with intuitive diagramming techniques. The central concept underlaying LabVIEW is the virtual instrument. The virtual instrument is similar in modularity and functions to a physical instrument; however, it is built out of software. This instrument is hierarchical, i.e., one virtual instrument can include a variety of other instruments or components. LabVIEW supports operating hierarchical instruments to perform complex tests and measurements in a manner similar to physical instruments connected with each other. Diagramming techniques give users opportunities to create a block diagram of an instrument and its front panel for interactive control using a computer mouse. Drawing an instrument diagram is directly associated with programming data acquisition, processing and displaying. That is performed entirely in a graphic programming language invented specifically for this task. That language uses data flow concepts, traditional program – control structures and a set of instructions consisting of graphical elements.

The signal processing functions in LabVIEW provide users with the most frequently used signal processing operations, e.g. FFT, fast convolution, power-spectrum analysis and many others. The FFT analyzer shown in Figs. 7 and 8 can serve as an example of a virtual instrument created using the LabVIEW system. The lay-out of that instrument was programmed and experimentally verified in the Music and Acoustic Laboratory of the University of Thessaloniki. Figure 7 shows the designed front panel of the virtual instrument. The program of that instrument written in the graphic language is presented in Fig. 8. The flow of data specified in the block diagram is created by selecting and arranging graphical objects and wiring them together. The FASTIO procedure performs very high speed input and output operations. It is specified by the parameters listed below: *trigger*, *points*, *loops*, *mult*, *device*, *type*, *offset*, *slot*, *error*. The *trigger* parameter determines the external triggering mode. Setting to zero indicates no external triggering is to be used, so the input or output operation begins immediately after calling FASTIO. The *points* parameter specifies the number of samples that are digitized. The number of data elements involved in processing is defined by the power of 2, where the range of power is assigned from 7 to 17. For input operations the *loops* parameter should be set to one. The *mult* parameter is used to determine the cycle time which corresponds to the duration between consecutive samples, and is calculated as *mult* $\times 0.2$ μs. It is determined by the computer being used and should not exceed the minimum and maximum time specifications which, in case of Macintosh II, are consequently 1.2 μs and 26 μs. The *slot*, *device*, *type*, *offset* parameters specify the installation of the

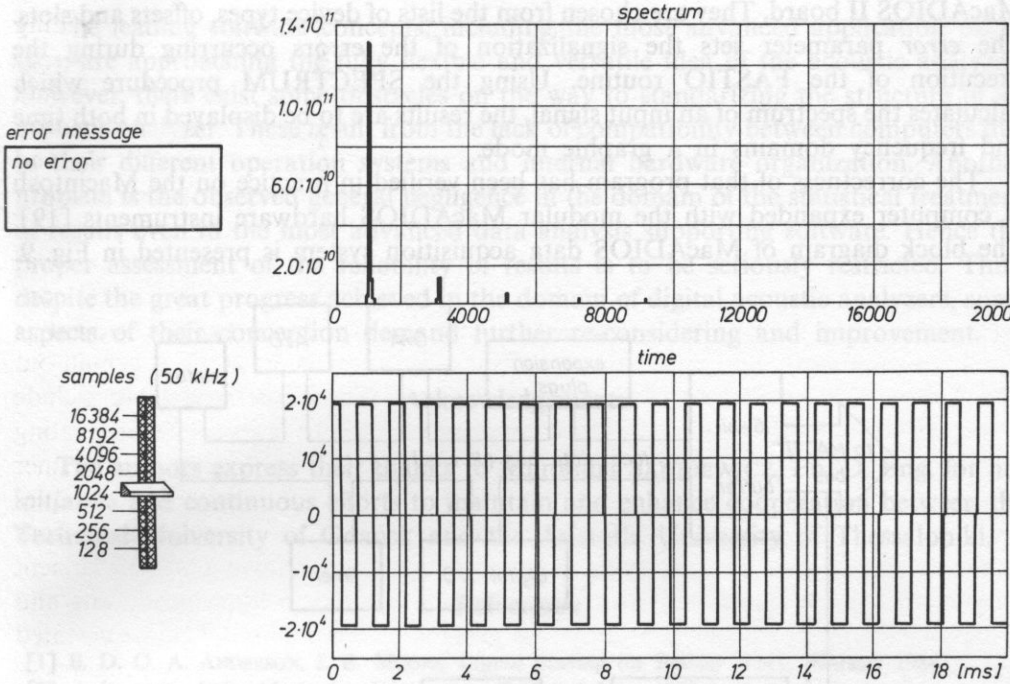

FIG. 7. Front panel of the FFT analyzer designed using the LabVIEW

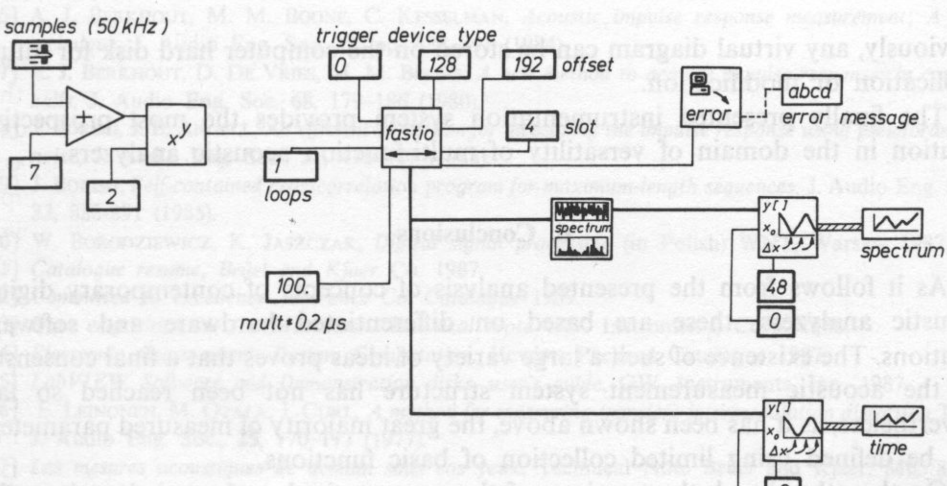

FIG. 8. Lay-out of the FFT analyzer

MacADIOS II board. They are chosen from the lists of device types, offsets and slots. The *error* parameter sets the signalization of the errors occurring during the execution of the FASTIO routine. Using the SPECTRUM procedure which calculates the spectrum of an input signal, the results are to be displayed in both time and frequency domains in a graphic mode.

The correctness of that program has been verified in practice on the Macintosh II computer expanded with the modular MacADIOS hardware instruments [19]. The block diagram of MacADIOS data acquisition system is presented in Fig. 9.

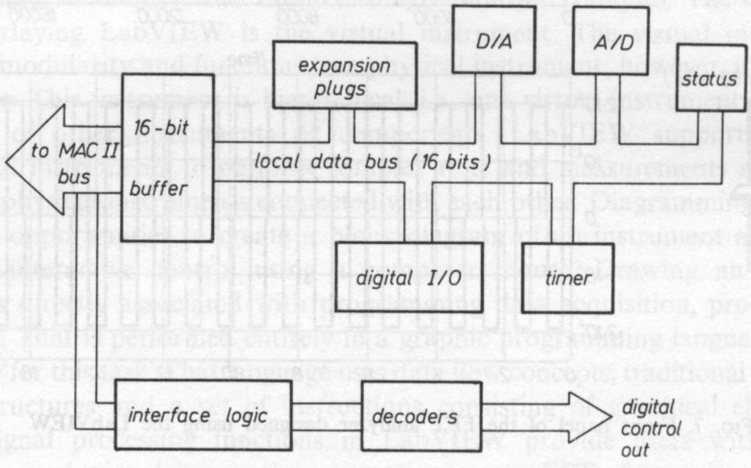

FIG. 9. Block diagram of the MacADIOS system

Obviously, any virtual diagram can be stored on the computer hard disk for future application or modification.

The finally presented instrumentation system provides the most prospective solution in the domain of versatility of multi-function acoustic analyzers.

6. Conclusions

As it follows from the presented analysis of concepts of contemporary digital acoustic analyzers, these are based on differentiated hardware and software solutions. The existence of such a large variety of ideas proves that a final consensus on the acoustic measurement system structure has not been reached so far. Nevertheless, as it has been shown above, the great majority of measured parameters can be defined using limited collection of basic functions.

On the other hand, the topology of the computerized analyzer is based on the restricted number of reconfigurable units responsible for the acquisition and processing of data. The two circumstances mentioned above lead to the modular conception of the analyzer in both hardware and software domains.

The leading software concepts, including the most advanced application packages, are approaching the fully flexible and versatile idea of the acoustic analyzer. However, there exist some obstacles on the way to standarizing the structure of an acoustic analyzer. These result from the lack of compatibility between computers due to their different operation systems and internal hardware organization. Another problem is the observed general negligence in the domain of the statistical treatment of results even in the most advanced data analysis supporting software. Hence the proper assessment of the reliability of results is to be seriously restricted. Thus, despite the great progress achieved in the domain of digital acoustic analyzers, some aspects of their conception demand further re-considering and improvement.

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