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ONE SOLUTION OF DIGITAL SIGNAL PROCESSING MODULE

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Abstract. Digital signal processing technique is used in many applications, some of them are data acquisition and processing in real time. For this purpose, numerous processing modules and boards are developed, ranging from simple developing modules to complex multi-processors systems. An unique module, which is designed and realized, is presented in the paper. The module is assigned for wide range of real time applications, involving waveform generation, signal classification and spectrum analysis. Examples of the module implementation in each of mentioned application areas are described.

Key words: Digital signal processing, multi processor system, waveform generator, spectrum analysis.

1. Introduction

The development of integrated circuits technology, together with the increase of implementation complexity, make possible the realization of programmable, integrated digital signal processors. They represent microcomputers specialized for digital processing of signals in real time. In general, the algorithms for digital signal processing (DSP) characterize intensive numerical calculations. Therefore, the architecture of digital signal processors should be considerably different with respect to conventional microprocessors in many aspects. Namely, they are distinct by drastic increase of numerical features in relation to standard processors which belong to the same technological generation [1]. Furthermore, the implementation of new technology in systems and devices realization, like the full digitalization and current signal processing methods, enables faster development of telecommunications.

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One of many applications where DSP is often used, are data acquisition and processing in real time. The architecture of conventional processors is not designed for fast and continual processing and memory access, which is indeed necessary for the real time operating. On the other side, this problem can be easily solved by using signal processors and for that purpose, numerous processing modules and boards are developed. They are used in wide range of applications, from simple developing modules, to complex multi-processors systems. Namely, certain multi-channel and very fast applications require more processing power from that is available in one processor, so it is indispensable to use several processors. The sampling rate, the channel number and the resolution are some factors which determine if one or more processors are required in given application. Parallel processing represents reliable method to accomplish the requirements of high speed and real time operating.

2. Choice of a Suitable DSP Module

A DSP module represent specific processor board, designed for data acquisition and processing in real time. It consists of one or more signal processors and additional hardware and it efficiently performs the specified mathematical operations, typical for signal processing. The DSP module is connected with a host. Usually as the host, a personal computer (PC) is used. The connection is performed on two ways: (a) the DSP module can be regarded as a separate unit, which communicates with the PC as an external peripheral unit; or (b) as an integral constituent of the PC hardware, while the communication take over one of the internal bus. There are currently several industry standard bus architectures in the PC compatible community: MCA, EISA, VESA, SCSI, ISA, PCI etc. Due to its appropriate technical features (32-bit bus, 33 MHz clock rate, transfer rate up to 132 Mbytes/s), the PCI (Peripheral Component Interconnect) bus is one of the most suitable for such high performance applications.

Besides the interconnection strategy, when the choice of suitable DSP module have to be made, it is important to evaluate other parameters, such as: the processor type, the module speed, memory capacity, the channels number, A/D converter characteristics, software tools, the price etc.

TMS320C3x and TMS320C4x processor families from "Texas Instruments" are mainly used for this purpose. One C3x processor type or more (2 — 4) C4x processors type are put into DSP modules, but there is an option with C4x processor and a vector processor, special kind of signal processors intended for vector and array operations. It is used in the case of spectral analysis, when high speed is required. Namely, vector processors can perform the FFT (Fast Fourier Transforms) operation more than ten times faster with respect to scalar processors. In a typical application, the vector processor is used for operations such as FFT, convolution or matrix computation, while the scalar processor performs auxiliary computation like digital filtering and makes decisions.

A memory for proceeded data storing can be local (SRAM type), when only one processor has access to it or shared, global ("dual port" DPRAM type), for two processors. Its capacity is chosen in respect to particular application and it can be up to 8 Mbytes.

In respect to the number of analog input channels, there are single-channel or multi-channel modules. In the second case, 4, 8 or 16 single ended channels are available. The main characteristics of an analog-to-digital (A/D) converter are the resolution and the sampling rate. The resolution usually corresponds to 12 — 16 bits. The sampling rate defines the highest frequency of signals that should be analyzed, while it takes values in the range 0.1 - 10 MHz.

Each hardware module is associated with comprehensive software support. It includes great number of DSP function libraries (FFT-s, filters, matrix calculations etc.) and various development tools like debuggers, compilers, Windows device drivers and interface libraries.

The prices of DSP modules are quite high. Simple modules cost few thousands US dollars, while complex modules are more expensive, up to 20000 $. The features that have the dominant influence on the price are the number of processors, the memory capacity and the sampling rate.

3. Examples of Available DSP Modules

Many DSP modules of various manufacturers can be found on the market, with different characteristics [2]. The module PC-430 from "Datel", which is connected with a PC via the ISA bus, is high speed DSP board. It contains a TMS320C30 40 MHz signal processor and 1 — 16 input A /D channels, with up to 10 MHz sampling rate. The card SPIRIT-40 PCI from "Sonitech", connected with a PC via the PCI bus, is a DSP system for multi-processing applications. It uses two TMS320C40 50/60 MHz signal processors, while it has single A/D channel with up to 10 MHz sampling rate or 8/16 channels with up to 750 kHz sampling rate.

The board SMT311 from "Sundance" is a powerful DSP module for performing vector algorithms on high bandwidth data. As a separate module, it communicates with a PC in a form of input/output unit. It contains a

TMS320C44 50/60 MHz signal processor and a LH9124 "Sharp" vector processor, and also four communication ports with 20 Mbytes/s transfer rate. Optimized for FFT operations, the SMT311 performs the real 2D 512$\times$512 FFT algorithm for 40 ms and the 1K complex FFT algorithm for 80 μs [3]. Similarly, a suitable system for high speed FFT-intensive applications is the module DSPC-49 from "PCM", with the ISA bus interface. It combines two TMS320C40 50 MHz signal processors with a LH9124 40 MHz vector processor, while it has two parallel (36-bit) ports and four communication (8-bit) ports.

The company "Innovative Integration" offers various modules, with the PCI interface. The module PCI32 is the most cost-effective DSP platform. It contains a TMS320C32 60 MHz signal processor and four A/D channels, with 200 kHz sampling rate. The module PCI44 is one of the fastest and most flexible system. It uses up to three TMS320C44 50 MHz signal processors, while it has single A/D channel with 10 MHz or 16 channels with 1.25 MHz sampling rate. The new products from this company are now available [4]. The module M44 is a high performance DSP board with a TMS320C44 60 MHz signal processor and flexible input/output interface. The module M62 is an ultra performance system with a TMS320C6201 133 — 200 MHz signal processor, which is capable to sustain computational throughput of 1600 MIPS (million instructions per second). These cards may be installed in a system with full driver support under Windows 95 and NT.

4. Basic Configuration

As it is mentioned, the DSP module prices are high. Because of that, a new solution is chosen and an unique DSP module is designed and realized for our special purposes.

The DSP module is assigned for wide range of real time applications. The module can operate as an autonomous device with addition of a keyboard and a display, or it can be an integral part of the system, where it is connected with a PC in a form of peripheral unit. The hardware platform together with software nature of the realization, enable simple manner of modifying existing or adding new functions, by changing the software. The module is realized as a 4-layer planar card, in 120 $\times$ 180 mm format. It contains: a signal processor, a micro-controller, A/D and D/A converters, RAM and ROM memories, clock generators 8 and 40 MHz, an address decoder, the RS232 interface and required control logic. Its basic configuration is shown on Figure 1.

The TMS320C50 signal processor from "Texas Instruments" is used for

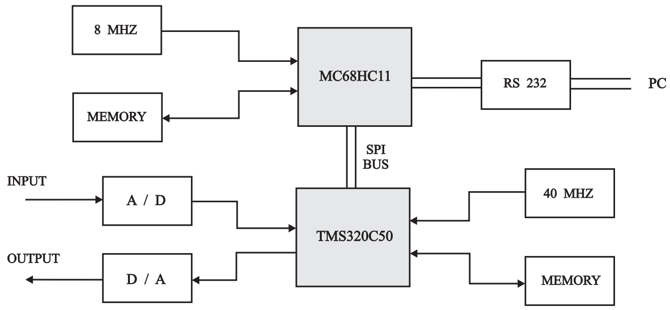

Fig. 1. DSP module basic configuration

carrying out all DSP algorithms. It is a static CMOS fixed-point, 16-bit processor, running at 40 MHz. The single-cycle instruction execution time is 50ns. Its architectural design is based on the combination of an advanced Harvard architecture (separate buses for program memory and data memory), on-chip peripherals and on-chip memory. Moreover, the TMS320C50 has highly specialized instruction set. These features enable the operational flexibility and the device speed, which together with the cost effectiveness makes this signal processor as the suitable choice for a wide range of applications.

The TMS320C50 has a programmable memory map (address range is 224K $\times$ 16 bit words, K = 1024), which can vary for each application. On-chip memory includes 10K words of the RAM and 2K words of the ROM. External memory on the DSP module includes 32K bytes of the erasable programmable ROM (EPROM) and 32K words of the RAM. User programs are stored in the EPROM, while for transferring this software to the program memory at power-up, a boot loader with parallel interface is provided. The RAM is used for data storing during processing.

The TMS320C50 can use either its internal oscillator or an external frequency source for a clock. The external divide-by-two option is used on the DSP module. For that purpose, the 40 MHz clock is externally generated with a high frequency crystal.

The DSP module control and supervision is carried out by a microcontroller. The MC68HC11 processor from "Motorola" is used for that purpose. It is an advanced 8-bit HCMOS microprocessor unit with highly sophisticated peripheral capabilities. The required 8 MHz clock is externally generated with a crystal. External memory on the DSP module includes 16K bytes of the EPROM and 8K bytes of the RAM.

The module can operate in two modes: autonomous and remote control. Adding the keyboard and display, the module becomes an autonomous device. Besides that, the module can be connected with the PC, in a form of input/output unit, by the RS232 interface, common, serial asynchronous interface. The communication between the microprocessor and the signal processor is carried out over the SPI (Serial Peripheral Interface) bus, serial synchronous interface.

For a conversion of digital signals in the analog form, the DAC813 converter from "Burr Brown" is used. It is a 12-bit digital-to-analog (D/A) converter, with a flexible interface that allows easy interface to the signal processor. The converter maximal resolution is 12 bits, maximal settling time is 5 μs and it is specified to $\pm 1/4$LSB linearity error and ± 0.02 offset error (maximum values).

For a digitalization of analog signals, the AD7891 converter from "Analog Devices" is used. It is an 8-channel, 12-bit data acquisition system with a choice of either parallel or serial interface structure. The converter maximal resolution is 12 bits, maximal sampling rate is 454 kHz, while its relative accuracy is $\pm 3/4$LSB in that case. So, the input signal bandwidth can be up to 227 kHz. The conversion clocks for both converters are obtained by dividing the signal processor external clock.

5. Implementation

The described DSP module can be used in wide range of real time applications. Using the TMS320C50 function of the boot loader (serial boot option), the special program for user software transfer from the PC to the DSP module is developed. This possibility enables the module operating as a developing tool. In addition, numerous software routines for the TMS320C50 signal processor are developed. On the basis of these routines and the module implementation, several devices are developed and realized. Application areas, related to radio communications, involve: waveform generation, signal classification and spectrum analysis.

5.1. Waveform generation

Radio communication systems are subjected to a wide range of interference. That is caused from many sources, including various forms of intentional or unintentional jamming and different kinds of noise. In order to evaluate the performance of radio equipment in presence of interference i.e. to simulate the real propagation conditions in laboratory environment, the need arises for a waveform generator. Such device has to be capable to produce various interference-like signals, enabling reproducible simulation tests to be carried out [5].

Using the DSP module, a waveform generator for producing different types of analog and digital signals is designed and realized. The following signals can be generated: the single tone, the Gaussian noise, the FM noise (in whole or limited range), the linear FM signal (in whole or limited range), the FSK signal, the electronic music and the pseudo random pulses. Each of them simulates another kind of interference. Computation and synthesis of these signals are carried out in the software manner, executing particular computer programs. This method provides fast, accurate and reliable synthesis of different waveforms and their parameters can simply be adjusted or changed. Optionally, the given signal set can be easily extended, without any hardware modifications. Waveforms are generated in the frequency range 20 — 25000 Hz, which is used for standard audio signal testing. The device can be easily incorporated into automated measurement systems.

The waveform generator basic configuration is given on figure 2. Its hardware realization is quite simple. The microprocessor provides local control (with the user interface) in the autonomous operation mode or remote control of the device in the remote control mode. The user interface module contains a LCD display and a keyboard, while in the remote control case, the microprocessor is connected with a PC using the standard RS232 interface. The signal processor performs the synthesis of desired signals. Generated waveforms have to be converted in the analog form and then they are adjusted for further processing in the output module, which includes a filter and an amplifier for analog signals (ANS) and a buffer for digital signals (DIS).

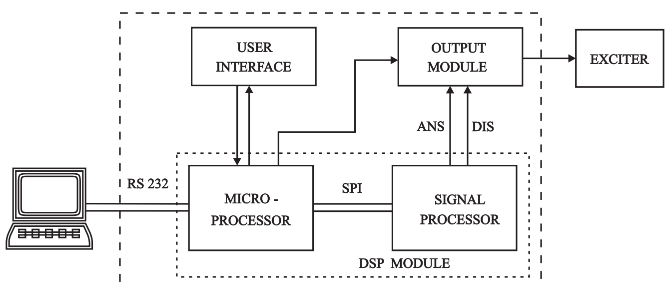

Fig. 2. DSP Waveform generator

For the synthesis of desired signals two software routines are developed:

- the sine-wave signal generation routine,

- the pseudo random sequence of maximum length 32 generation routine.

Few algorithms for the sine-wave signal generation exist, while the linear interpolation method is chosen as the most suitable among them [6]. It is based on reading the sinusoid memorized samples from a look-up table and using an interpolation scheme to obtain more accurate signal samples. The values of the sinusoidal function for N uniformly spaced angles, into the range $0 - 2\pi$, are stored in a table. The sine-wave signal is generated by stepping through the table at a constant rate and wrapping around the end of table whenever 2π is exceeded. To compute the sine values between table entries more accurately, a linear interpolation scheme is implemented.

The pseudo random binary sequence (PRBS) of maximum length 32 generator can be realized as a shift register with proper feedback circuitry and arbitrary initial state. Combining these two routines, all desired waveform types are generated. For each signal, a particular program is developed. So, the sufficiently quick and accurate calculation for complex signal forms is provided.

The sine-wave signal generation routine is used for the single tone synthesis. The resolution of generated tone frequency amounts $\Delta = 0.763$ Hz.

The idea of adding one-bit numbers, submitted to the uniform distribution is used for the Gaussian noise generation. Namely, according to the well-known "Central Limit Theorem", the Gaussian distribution of signal amplitudes is expected if sufficient number of uniformly distributed amplitudes are added. For this purpose, a pseudo random generator can be used, while the given PRBS generation routine is implemented for its realization. Thus, the one Gaussian noise sample is obtained by adding ten successive generator states.

The FM noise contains single tone series, randomly chosen from a given frequency range. Each tone is generated during specified constant time period, using the sine-wave signal generation routine. For the choice of particular tone, the PRBS generation routine is implemented. Namely, the next frequency is defined on the basis of pseudo random sequence state.

The linear FM signal represents a tone, which frequency is linearly changed in the given range. The whole assigned range have to be swept during the required time interval, while the unit value is chosen as frequency increment and decrement. So, each tone is generated during specified time period, which duration depends of the sweeping range width. The modified sine-wave signal generation routine is used for this complex signal generation.

The FSK signal consists of two tones, which are randomly generated during specified time interval. Both tones are generated using the sine-wave signal generation routine, while the PRBS generation routine is used for the tone choice, such that the one specified bit (its logic state) from the binary sequence defines which tone should be generated.

The electronic music presents a sum of two or more single tones. For the generation of this complex waveform, the modified sine-wave signal routine is used.

The pseudo random pulses present series of pulses, whose logic state is changed according to the pseudo random law. For the pulse generation, the PRBS generation routine is used, selecting one, specified bit from the pseudo random sequence, which is brought to the separate, digital output.

5.2. Signal classification

One of the main functions of electronic surveillance and spectrum monitoring systems is recognition of radio emissions on the basis of their modulation type. Signal classification is very complex task, especially in multi-signal environments, like HF, VHF and UHF frequency bands, where mostly current civilian and military radio communication systems operate. Even experienced radio operators are often unable to analyze and identify differently modulated signals, in real time, with a help of conventional devices. It is happen because of high signal density, as well as existing of fading effects and short emissions in the same time in given ranges.

A general unifying concept for solving the problem of radio emissions type recognition does not exist. For the implementation in mobile surveillance systems, the automatic recognition of specified analog and digital modulation types in given range, in real time, with high probability of signal classification, is required. One such device is designed and realized, using the DSP module. It allows automatic recognition of most AM and FM emissions, which are usually met in HF, VHF and UHF bands. The signal classification procedure is based on the method of signal envelope and instantaneous frequency detection [7]. It represents a simple feature extraction and processing algorithm, which uses the fact that information about AM and FM modulations are contained in the signal envelope and its instantaneous frequency. Several signal features, named separation parameters, are extracted from them. Calculation and processing of these parameters give the possibility to classify signals according to the modulation type. This procedure is realized in a form of two independent programs. The software nature of proposed solution enables to achieve the realization of identical devices. In addition, the problems about adjusting and checking of analog components and parameters are avoided. Finally, the development of a complex classification algorithm is enabled, which provides more reliable and accurate results.

The signal classifier basic configuration is given on figure 3. Similar as for the waveform generator, the device can operate in two modes, autonomous and remote control. Control functions are provided by the microprocessor, together with the user interface, a PC and the RS232 interface.

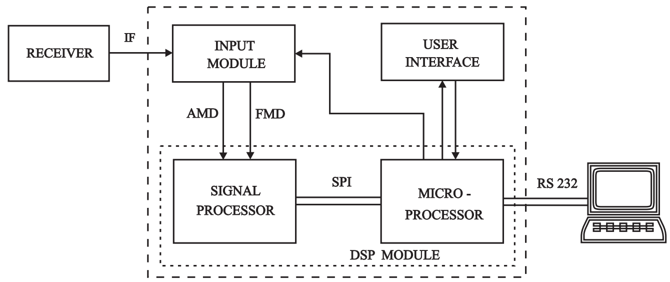

Fig. 3. Signal classifier

Before starting the classification, modulated RF signals are preprocessed in conventional, digitally controlled radio receiver. The preprocessing includes: signal conversion to intermediate frequency (IF) 455kHz, filtering and automatic gain control, enabling the classificator performance independently of the signal level. The classification procedure consists of three processing stages. In the first stage the IF signal demodulation is carried out in the input module. An envelope detector is used for obtaining the AM demodulated signal (AMD), while a FM discriminator is used for obtaining the FM demodulated signal (FMD). In the second stage these signals are converted into a digital form, then they are processed and the separation parameters are generated. For this purpose, an algorithm for the signal processor is developed. Finally, in the third stage the decision about of modulation type is made, on account of the separation parameter values and predetermined criteria. For that purpose, an algorithm which performs the microprocessor is developed. It is based on numerous measurements of the separation parameters values for both, test and real signals. On the basis of described method, the automatic recognition of following modulation types is enabled: the noise, the unmodulated carrier, the KAM voice, the SSB voice, the binary ASK, the binary FSK, the multichannel FSK, the analog FM and the digital FM.

The selected signal features extracted from the signal envelope are activity factor (the mean value of detected signal envelope), AM variations (the signal envelope dispersion) and crossing rate (the number which shows how many times the envelope crosses specified level, 2 threshold levels exist), while features extracted from the instantaneous frequency are narrow band and wide-band FM variations (the mean value of filtered and two-wave rectified signal, different filters are used). Therefore, six separation parameters are deduced from these extracted features. In order to calculate desired separation parameters, the special algorithm is developed. It is based on three software routines:

- digital filtering,
- average value calculation,
- a dispersion calculation.

The routine for digital filtering is used to realize low-pass, high-pass and band-pass filters. The filter type and its parameters are set with previously calculated coefficients. The signal mean value is calculated as a mean value of n envelope samples $\left(\sum x\right)/n$, while the dispersion of n envelope samples is calculated using the relation:

$$\text{dispersion} = \sqrt{\frac{\sum x^2}{n} - \left(\frac{\sum x}{n}\right)^2}$$

Using these routines and the particular program for crossing rate counting, whole separation parameters can be easily determined.

5.3. Spectral analysis

In the classification procedure, large number of collected data about detected signals appear. During the analysis, it is often required to monitor them. The most suitable method of monitoring is the graphical presentation, since it enables to users the visual survey of results. It was rather hard to accomplish this task, until nowadays. However, with the appearance of new multitasking operating systems and advanced developing tools, the graphical presentation of collected data has became quite simple. Thus, current visual program languages enable the easy creation of graphically oriented user interfaces and the function of detected signals monitoring can be provided.

In order to achieve the required option of monitoring, an otherwise solution for signal classification is proposed. It is completely digital method, while analog components are used only for converting the receiving signal in desired frequency range. The essential conception is that the classification procedure is carried out in frequency domain, based on the recognition of signal spectral shapes. Proceeded data are further transferred to a PC, in order to memorize and monitor them. The monitoring with panoramic data display function can be simply added, by modifying the software. This function is used for the spectrum occupancy analysis.

The suggested solution for automatic signal classification and monitoring with panoramic data display is realized as one module within a surveillance system [8]. The basic operational principle is shown on figure 4. For signal detection, a wide-range, radio surveillance receiver is used, with remote control capabilities. The received signal is converted from IF 455 kHz to the base band 0 — 20 kHz, while it remains modulated. The signal bandwidth of 20 kHz is sufficient for all desired signals that should be classified. After that, the signal is brought to a compression amplifier, to adjust its level before A/D conversion. Then, the signal is directed to the DSP module. The DSP module main functions are: the received signal digitalization, the conversion from time to frequency domain, the signal classification and the movement of proceeded data to the PC.

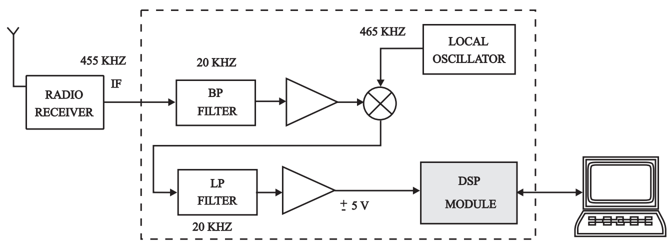

Fig. 4. Spectrum generation and monitoring

After the received analog signal conversion in a digital form, it should be transformed in frequency domain, calculating Discrete Fourier Transform. For this purpose, the Fast Fourier Transform (FFT) algorithm is applied. Since the spectrum of received signal is produced, it is possible to recognize its modulation type. Knowing the spectral shapes of desired signals and according to predetermined classification criteria, the decision about modulation type can be made. Then, data about the received signal spectrum and classified emission type are sent to the PC, which is free for executing its jobs. The required real time operating is achieved on that way.

Obtained data about the signal spectrum and classified emission type are stored in the PC memory. Due to the spectrum occupancy analysis, generated spectrum samples can be graphically presented on the PC monitor. The monitoring function is achieved using the Windows NT operating system and the Visual Basic program language. Different display forms are available to the user. The panoramic display in whole required range (1.5 MHz) or its particular, narrow part is available.

For the spectrum generation of detected signals, the unique, complex FFT algorithm, up to 8192 points, is realized. It represents modified version of the 1024-point (1K $\times$ 1K) complex FFT software package [9]. In the case when the spectrum is monitored in whole required range, the display range amounts 1.5 MHz and it is set by the start and stop frequency. The 16-point (16 $\times$ 16) FFT algorithm is used for that purpose. In respect of the signal bandwidth (20 kHz) defined with the bandwidth of used filter (figure 4.), the resolution of obtained spectrum is 2.5 kHz, what is entirely enough according to the given display range. Simultaneously, high speed of the FFT algorithm execution is enabled, which is very important because of large number of the FFT algorithm repetitions indispensability. Namely, on account of the required display range and the signal bandwidth, there are 75 channels and hence the FFT algorithm have to be repeated 75 times. The speed of spectrum generation is about 20 channels/seconds. As an example of this wide-band display method, the signal spectrum in the range 187 — 188.5 MHz is shown on Figure 5.

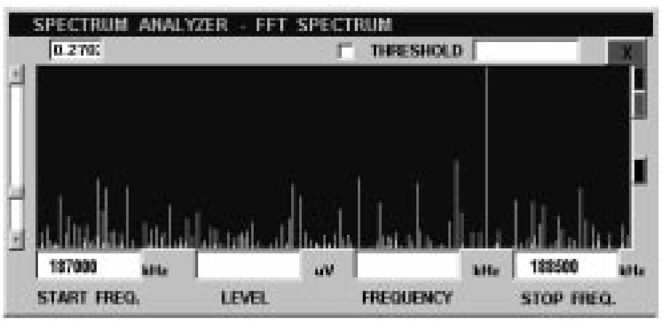

Fig. 5. Generated FFT spectrum

When more detailed analysis is required, the narrow-band display method is implemented, while 20 kHz bandwidth is considered. For this purpose, the 1024-point (1K $\times$ 1K) complex FFT algorithm is applied. Because of that, the spectrum resolution is much better and it is around 40 Hz. The narrow-band display can be used for those signals selected during the wide-band display or for signals assigned from a certain list, when their presence is expected in advance.

6. Conclusion

DSP is used in large number of applications and data acquisition and processing in real time are some of them. For this purpose, numerous processing modules and boards are developed, ranging from simple developing modules to complex multi-processors systems. An unique DSP module, which is designed and realized, is presented in the paper. The module is assigned for wide range of the real time applications, involving signal generation, signal classification and spectrum analysis. As examples of the module implementation, the realized devices for each of mentioned application areas are described. The DSP module hardware platform, based on the signal processor and the microprocessor, together with software nature of the realization, gives many features like: simplicity of realization, fast and reliable computation, implementation flexibility, the full repetition possibility, easy modification of existing or addition of new functions, small dimensions, low cost.

REFERENCES

1. M.Popović: *Digital Signal Processing.* Nauka, Beograd, 1994.
2. V.Matić, B.Pavć, V.Tadić: *The implementation of digital signal processing for automatic recognition of radio signals type.* JISA INFO, no. 2–3, pp. 45–50, 1999.
3. Innovative Integration Product catalog: *1998.*
4. Sundance SMT311 Press Release: *1996.*
5. B.Pavić, V.Marinković: *Coprocessor-based waveform generator for laboratory testing of radio systems.* Conference on New Trends in Communications, Control and Signal Processing, Ankara, 1990, Proceedings, pp. 590–595,.
6. V.Matić, V.Tadić, V.Marinković: *Comparison of digital signal processing methods for sine wave signal generation.* Symposium ITS'98, Sao Paulo, 1998, Proceedings, vol. 1, pp. 290–295.
7. Z.Krsmanović, P.Petrović, N.Remenski: *Automatic classification of HF radio signals.* Global Communications, IV quarter, pp. 33–36, 1989.
8. V.Matić, B.Pavić, V.Tadić: *Automatic recognition of radio emissions type and spectrum occupancy analysis using a personal computer.* Conference YU INFO'99, Kopaonik, 1999.
9. K.Lin: *Digital Signal Processing Applications with the TMS320 Family.* Prentice-Hall, New Jersey, 1987.