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Implementing a Real Time System for Lung Sound Spectrographs Using the TMS320C30 DSP

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EFRIE, France

December 1995
SPRA310

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Abstract

This application report describes how to develop a real-time system for producing lung sound spectrographs and other analyses using the Texas Instruments (TI™) TMS320C30 (C30) digital signal processor (DSP). The C30 floating point processor, hosted by a PC with a 486 DX2 processor (or better) is written entirely in the C language. With this real time system, analog input can be captured directly from a microphone and processed on the C30. The results can be sent to the PC and displayed graphically with negligible delay.

This document was an entry in the 1995 DSP Solutions Challenge, an annual contest organized by TI to encourage students from around the world to find innovative ways to use DSPs. For more information on the TI DSP Solutions Challenge, see TI's World Wide Web site at www.ti.com.

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Introduction

The use of digital signal processing (DSP) techniques to analyze pulmonary sounds is well established. PC-based systems have been developed to perform various types of time-domain and spectrographic analysis.[1 2 3 4 5 6] The availability of high-speed microprocessors specifically designed for digital signal processing has been exploited in the development of such systems.

Recent improvements in the speed and power of these special purpose processors have greatly increased possibilities for the system designer. This application report describes the development of a system for producing lung sound spectrographs and other analyses in real time.

Real time analysis refers to the analysis of input signals as they are captured. The results are displayed in graphical form with negligible delay on the PC screen. We believe that real time analysis has the following advantages:

- Analysis is fast, apparently instantaneous.
- Useful data can be immediately identified, thus making it easier to record only characteristic features of interest. This eliminates searching through lengthy computer files to find such data.
- Storage requirements are reduced.
- Errors in the recording equipment or the environment are easily rectified before analysis begins, and before any conclusions are drawn or permanent records produced.
- Adjustments can be made to amplifier gains, analog and digital filters, microphone placing, the sampling rate, etc. The immediate effect is seen on the computer screen. These settings can thus be optimized.
- Different observations about the patient are correlated with the display.
- The effect of changes, for example due to treatment or challenges, is observed and measured as changes occur.
- Possible confusion and misinterpretation of results are reduced or eliminated.

The TI TMS320C30 floating point processor is hosted by a standard PC with 486 DXS processor(or better) and entirely written in the C language.[7] Details of the system are presented with examples of the clinical research that has influenced its design.

Description

Analysis

The primary requirement in this work is the production of a real time color spectrograph of lung sounds obtained from a microphone, along with a calibrating signal obtained from a pneumotachograph indicating airflow level.

The color spectrograph is a three dimensional graph displaying successive segments of the microphone signal with frequency on the vertical axis and time on the horizontal axis. Color is used to indicate the spectral energy in a given frequency range at a given point in time. Bright colors indicate high energy levels and dark colors indicate low energy.

The spectrograph is obtained by a succession of fast Fourier transform (FFT) analyses performed on Hamming-windowed segments of suitable length. The real time processing requirement means that the FFT and display processes for a segment must be complete in less time than it takes to capture the next segment.

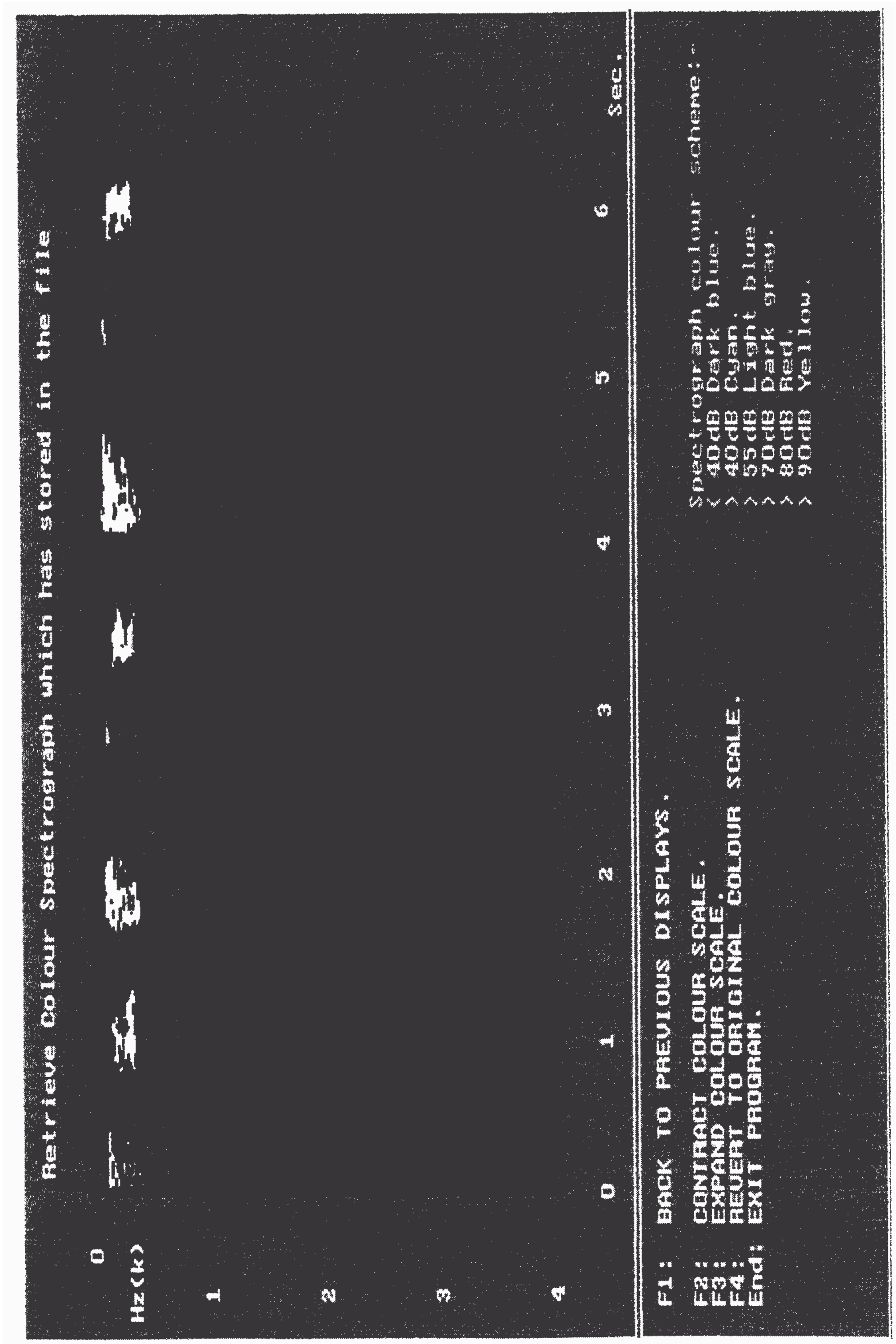
For example, with a sampling rate of 8 kHz, (and therefore a signal bandwidth of up to about 4 kHz) and a segment length of 256 samples, the FFT and display process must take less than 32 ms. This is well within the capability of floating point DSP processors such as the TMS320C30 linked to modem personal computers.

An additional requirement for this work is the production of a real time linear prediction (LP or LPC) model that can produce less complex spectrographs emphasizing features of interest.[6] Such a model can also be interpreted in terms of a lossless acoustic tube with dimensions reflecting those of the human vocal tract when excited by a sound source.

For voiced speech, the sound source would be the vibrating vocal cords and additional use is made of LP or LPC models in speech processing research.[8] Such models have also been used in the study of snoring and sleep apnea where the sound source occurs at some obstruction in the upper airway, such as the soft palate.[9]

The real time analysis system described in this work produces FFT and LP spectrographs and a lossless tube representation of the LP model.

Figure 1. Lung Sound Spectrograph



Equipment

The system uses a TMS320C30 processor situated on a relatively simple card called an evaluation module (EVM) card manufactured by TI. The card is connected to the ISA bus of a 486 DX2 PC. The EVM card has analog input and output connections, programmable anti-aliasing filters, and 14 bit analog-to-digital (A/D) and digital-to-digital (D/A) converter.

Analog signals are input directly from a microphone into the card and processed by the TMS320C30. Results may be sent to the PC. The PC communicates with the EVM via a bi-directional communication port that allows programs to be downloaded to the TMS320C30. It also allows data to be interchanged between the EVM card and the PC while programs are running.

The C30 DSP device post-dates first and second-generation DSPs with 16-bit fixed-point arithmetic introduced in the mid-1980s. The third-generation C30 is a 32-bit processor with floating point arithmetic capability. Its four levels of pipelining mean that as one instruction is executed, the next three are fetched, decoded, and read.

Because instructions can be executed in parallel, it is possible to execute up to 33 million instructions per second. The architecture and instruction set are specifically designed for the type of operations commonly expected in DSP procedures. Many special features, such as bit-reverse addressing for example, can be usefully exploited.

Software

The software required to produce and display a real time FFT color spectrograph consists of two programs, which run simultaneously and intercommunicate. One program runs on the TMS320C30 and the other runs on the PC. Both programs are written in C language.

The TMS320C30 program was converted to TMS assembly language using a cross-compiler supplied by Texas Instruments. Details of the programs are given below:

TMS320C30 Program

The TMS320C30 program:

- Initializes the EVM system, sets the frequency of the sampling rate timer, and selects the cut-off frequencies of the anti-aliasing analog filters.
- Enters an infinite loop with three concurrent processes:

- Process 1

  Under interrupt control, capture new blocks of input samples from the A/D converters

- Process 2

  Process the previously captured blocks of input samples to produce processed blocks

- Process 3

  Using direct memory access (DMA), transmit the previously processed blocks to the PC

Therefore, processes 1 and 3 continually interrupt the analysis performed in process 2. The sampling rate is fixed at 8 kHz to accommodate lung sounds of bandwidths up to about 4 kHz. Therefore, the input sampling process interrupts the other processes once every 125 µs to store an input sample into the new block being constructed.

The response to each interrupt takes only about 1 µs of processing time. Array pointers provide a very efficient way of transferring data from the input blocks, when full, to the analysis blocks. Instead of transferring data from one array to another, pointers to the arrays are simply swapped as follows.

There are three arrays (*A*, *B*, and *C*) and three pointers (*input*, *output*, and *intermediate*). The array pointed to by the input pointer receives data from the A/D converter. The intermediate pointer points to the array being analyzed. The *output* pointer points to the array being transferred by DMA to the PC.

Initially, the input, output, and intermediate pointers point to arrays A, B, and C, respectively. When the input array is full, the input pointer is changed (for example, from array A to array B) the output pointer is changed (for example, from B to C), and the intermediate pointer is changed (for example, from C to A). Therefore, what was previously the input array becomes the intermediate array and is subject to analysis.

The previous intermediate array now becomes the output array, assuming the required analysis is complete and the results are stored in this array. The contents of the previous output array assumes the transfer to the PC, and therefore the previous output array becomes the new input array to be filled again with input samples from the D/A converter. This pointer exchange occurs each time that the input array is filled.

Communication from the output array to the PC takes place by DMA. With this technique, a separate controller that works in parallel with the TMS320C30 without affecting its operation transfers samples. It is necessary for the C30 to specify the source and destination memory addresses, specify the number of words to be transferred, and initiate the transfer.

The EVM card uses a bi-directional port to realize the communication. This is a combination of two ordinary registers. For each register the latch control and the 3-state control are connected to different processors. For one, the latch control is connected to the PC and the 3-state control is connected to the EVM. For the other, it is just the opposite.

Thus, data from the PC can be written to a register appearing as an output port and read from the same register by the EVM because the same register appears to it as an input port. This works the same for communication from the EVM to the PC. As far as the PC is concerned, the latch control and 3-state control are controlled by one address. The distinction is made by the read and writes control line. On the EVM side again there is only one address.

PC Program

Compared to the TMS320C30 program, the PC program is relatively simple and includes the following tasks:

- Download the EVM program and establish inter-communication
- Send commands to the EVM requesting data transfer and receive blocks of data for display or storage
- Display the color spectrograph and/or time waveforms on the screen

These tasks must be carried out simultaneously with the three tasks being executed by the TMS320C30.

Display

There is a range of different analysis and display options. The real time color spectrograph is a 4 kHz, or 1 kHz bandwidth.

Alternatively, a single changing power spectrum or time-domain waveform display may also be selected. For every display, for example, several controls allow displays to be held, or *frozen*, stored on disk, retrieved, and/or plotted with a different color scale.

A process of decimation obtains the 1 kHz bandwidth displays. For the 1 kHz bandwidth spectrograph, the input-sampling rate remains at 8 kHz, and the input anti-aliasing filter is reprogrammed to cut-off at about 1 kHz.

A digital anti-aliasing filter that improves the suppression of frequency components above 1 kHz supplements the effect of the filter. The sampling rate is then reduced to 2 kHz by omitting alternate samples. The block length, thus reduced by a quarter, is restored to its original value by *zero-padding*, that is, by appending three-quarters of a block of consecutive zero-valued samples.

The time-scale for the 1 kHz spectrograph is therefore the same as for the 4 kHz spectrograph, though the effect of the zero padding is to produce a more smoothly interpolated spectrograph.

Real Time Spectrograph Application to Clinical Research

The system described in this application report was developed for clinical research with objectives that include the following:

- An increase in our understanding of the origin of normal and abnormal lung sounds
- Discover how and to what extent computerized DSP systems could help with the diagnosis of chest diseases
- Find new ways of facilitating the management of chest disease

This technology has opened up exciting new horizons. Presented below are two examples of how the spectrograph has helped us in our understanding of chest disease.

A series of experiments described in a work soon to be published have improved our understanding of the mechanical events associated with wheeze production.[10] Although wheezing heard at the mouth or with a stethoscope in an asthmatic patient is a classical sign of airflow obstruction, relatively little is known about the relationship between the sound and the site and mechanism *of* its production.

In normal people, such airflow limitation and wheeze only occur in forced expiration but in asthmatics it can occur in normal breathing. In the study we used methacholine to produce airway narrowing in eight susceptible subjects until either the onset of wheeze or the occurrence of other significant symptoms. The investigation was performed in a constant volume body plethysmograph to provide detailed measurements of lung mechanics. In particular, tidal and maximal flow loops were produced to indicate the presence of airflow limitation.

Breath sounds were detected using an air-coupled electric microphone fixed over the right upper chest by a suction cup during quiet tidal breathing. We detected wheezing using the spectrograph. It was observed that:

- Wheeze was of abrupt onset and once established was heard in every breath
- Expiratory wheeze occurred late in the challenge
- Expiratory wheezes were not related to peak expiratory flow but were associated with the flow-limited part of the flow-volume curve
- All patients with expiratory wheeze had evidence of tidal airflow obstruction during the challenge and this was usually well established by the onset of wheeze. Thus expiratory wheeze seemed to be associated with airflow limitation.
- Expiratory wheezes all decreased in frequency during the challenge
- During inspiration, wheeze occurred when flows were maximal and there was no FJV loop evidence of airflow obstruction. The changes in frequency in inspiratory wheeze were inconsistent.

These observations lead us to conclude that:

- Expiratory wheeze appears to be associated with expiratory flow limitation although the trans-pulmonary pressure required to induce airflow limitation may sometimes be insufficient to induce a wheeze, a further increase in pressure being necessary to deform the airways sufficiently to generate a sound.
- Inspiratory wheeze appears not to be associated with inspiratory flow limitation.
- The changes in the dimensions of the intra-thoracic airways that occur during breathing may explain the poorly understood alteration in the pitch of wheeze during expiration. In expiration, airways are likely to shorten and increase in density, which reduces the pitch of a wheeze.

A further illustration of the use of spectrographic analysis concerns its application to the lung sounds produced by a patient with the rare condition called *Tracheomalacia*. Tracheomalacia causes the airways to lose all their cartilaginous support. We noted a low-pitched vibration during expiration that we considered to be produced by the large airways fluttering. The flow trace clearly showed this oscillation and the spectrograph was used to analyze the sound.

The wide-band spectrograph had clearly identifiable high-energy bands up to 500 Hz. The reduced bandwidth spectrograph (0 to 1 kHz) displayed, within each breath cycle, one second episodes for which five or six strong periodic energy bands (with a harmonic structure) were observed with a fundamental frequency of about 50 Hz. They were of sudden onset in early expiration and the pitch frequency decreased at a rate of approximately 10 Hz per 200 ms.

In addition, the harmonic structure also altered with time. These findings are truly in keeping with the fluttering of a collapsible floppy tube. The measurements enabled us to make some useful observations about the patient and the nature of flutter in floppy tubes. When such flutter occurs in smaller airways this is thought to produce wheezing.

As well as identifying individual spectrographic features, other information can be extracted from this method of lung sound analysis. Series of spectra can be averaged and the aggregated spectral energy plotted. From such plots of aggregated power the frequency below which 25, 50, 75, and 95 % of the power occurs can be computed and such results have been used to determine the quantitative changes in lung sound energy during histamine and methacholine challenge.

Summary

This project presents technical details about a real time spectrographic analysis system specifically designed for the study of lung sounds, and briefly illustrates some of its current applications. The real time aspect undoubtedly makes it easy to use as we can now examine spectrographs and other representations apparently instantaneously, modify the capturing and processing parameters as needed and immediately identify useful data.

There is no doubt that an enormous amount of information is present within such spectrographs. However, it is also clear that further research is needed to identify and extract characteristic features, and there is much scope for further work in this area.

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