Implementing an In-Service, Non-Intrusive Measurement Device in Telecommunication Networks Using the TMS320C31

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Abstract

Telecommunication channels cannot be monitored in an operational mode with traditional instrumentation systems. For example, transmission anomalies are usually not detected, which affects the quality of service to the end user.

This application report describes the implementation of the Texas Instruments (TI™) TMS320C31 digital signal processor (DSP) as an in-service, non-intrusive measurement device (INMD). The TMS320C31 can be connected to any easily accessible test point of a telecommunication network. The DSP-base unit provides interface circuits that allow operational mode monitoring in which transmission anomalies can be assessed for quality of service to the end user.

This application report includes a simplified telecommunication channel model, a summary of the hardware architecture, the adopted estimation algorithms, a brief description of the operating system kernel, and simulation results showing the DSP-based unit accuracy.

This document was an entry in the 1996 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

In the last few years, INMDs have assumed a growing interest because of the ability to monitor telecommunication channels in their operating mode. Thus, the effective degree of quality of the service offered to the end user can be assessed and transmission anomalies, which are usually not detectable with traditional instrumentation, can be observed.[1]

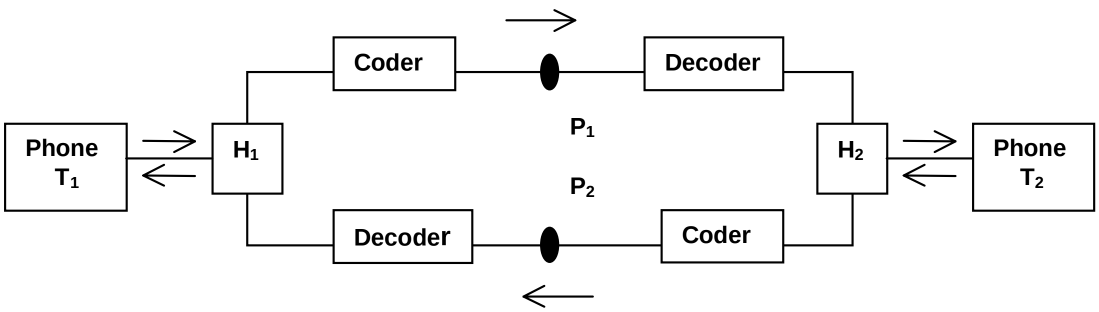
The DSP-based unit described in this document is connected to any easily accessible test point of the telecommunication system. The adopted DSP is provided with interface circuits that allow the acquisition of the digital data stream of modern communication networks. Using numerical algorithms, the system simultaneously performs a number of measurements in compliance with international recommendations, including noise and signal level estimation, echo parameter extraction, and voice-data discrimination.[1] The DSP-based device can be used as a stand-alone unit or as the field unit of a more complex network supervision system. In this second form, it can exchange data with a remote host computer that controls more DSP units and collects the measurement results in a database. Furthermore, the host unit allows the presentation to the end user of a number of statistical indices related to the data obtained throughout the various measurements.

A description of the DSP-based sub-unit is provided in this document. The hardware architecture is summarized and the various measurement algorithms are quoted and discussed as to how they cooperate to give simultaneous measurements on various communication links.

Simplified Model of Telecommunication Channels

For non-intrusive measurements, a telecommunication channel can be modeled as shown in Figure 1.

Figure 1. Simplified Model of a Telecommunication Channel



The analog signals, generated by either of the devices T_1 and T_2, are at first conveyed by a short-distance, bidirectional, analog two-wire line. The analog signals are then separated by the proper hybrid circuit (H_1, H_2), filtered using a band-pass filter with cutoff frequencies equal to 0 and 4 kHz to avoid aliasing, sampled at F_s = 8 kHz, and quantified by a uniform 13 bit or 14 bit quantifier.

The quantified signals are compressed and encoded on 8 bits according to the A-law or μ-law pulse code modulation scheme. Finally, the encoded signals are transmitted on unidirectional digital channels into the four-wire loop. In this way, each sample of any of the quantified signals is expressed by means of an 8-bit A-law or μ-law PCM code. Hence, since F_s = 8 kHz, for each unidirectional digital channel a 64 Kbit/s binary data stream is generated.

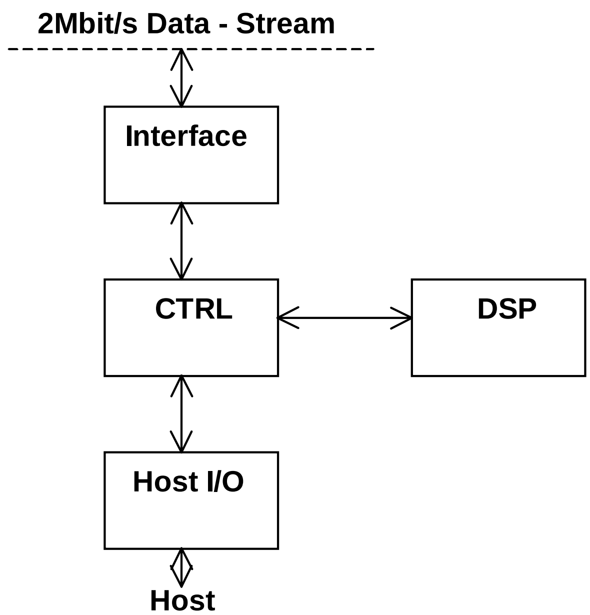
In modern European telecommunication systems, 32 64 Kbit/s data streams are time-division multiplexed thus generating a 2 Mbit/s data stream. In this way, each sampling period of duration T=125μ*s* is divided into 32 equal duration time slots. Each time slot is associated to one of the 32 time division multiplexed channels.

According to the CCITT European Standard, one of the 32 time slots is used for synchronization purposes and one for common signaling. Hence, only 30 voice-band channels are actually available.

The DSP-Based Instrument Hardware Architecture

A simplified block scheme of the DSP-based Instrument is shown in Figure 2.

Figure 2. Block Scheme of the DSP-Based Unit



The proposed system allows the monitoring of two full-duplex voice band channels, corresponding to four time slots in the time-division multiplexed PCM frame out of the 32 simultaneously transmitted.

According to international recommendations, the INMD can be connected only at four-wire test points, such as P_1 and P_2 as shown in Figure 1. The system interacts with a host computer using a modem through the telephone network in a rather complex way. Successive operations are required for the acquisition of the correct samples of the observed channel from the data stream. For this reason, a general-purpose microprocessor has been used, which is represented by the CTRL block of Figure 2. The CTRL unit also controls a switch matrix that can connect the selected channels to the I/O serial port of the DSP.

The HOST I/O block is constituted by a modem that allows transmitting and receiving data and commands through the telephone network. In this way, a single host computer can talk with several remote DSP based units.

The DSP unit that constitutes the core of the instrument is based on the TI TMS320C31, a 32 bit-floating point processor with a single-cycle execution time equal to 60 ns in the selected configuration.

The TMS320C31is includes

- Two 2 Kbyte blocks of internal RAM
- One 512 Kbyte block external static RAM
- One 512 Kbyte block of external dynamic RAM
- I/O port
- Direct memory access (DMA) controller

The DSP acquires a 32 bit word via a DMA channel every 125 μs. In each acquired word, the first 8 bits represent the A-law PCM coded sample corresponding to the first selected channel, the second 8 bits represent the sample corresponding to the second selected channel, and so on.

In this way, four data buffers are generated, each one corresponding to one of the four selected channels. The data buffers are then concurrently processed according to the algorithms briefly summarized in the following section, *Measurement Algorithms*. The results are written in dual-port memory 512 Kbytes deep, which can be accessed both by the DSP and the CTRL block.

Measurement Algorithms

To characterize the degree of quality provided to the user by a telecommunication system, a number of parameters should be evaluated during a telephone call. These parameters usually well describe the degree of quality of a voice communication, but assume a minor interest or a wrong meaning when data communications are involved.

Moreover, the percentage of data communications out of the total (data-traffic) is a meaningful and useful parameter when monitoring a telephone-type network because it helps the network provider upgrade the communication channels according to the needs of the user. Therefore, the first operation to be performed is the voice-data discrimination. To this aim, a number of statistical indices are evaluated from the signal. Then, a Bayesian classifier opportunely trained takes the decision by comparing such indices with a threshold chosen in such a way to minimize the probability of classification error.

If the communication is a voice communication, a further analysis is performed to detect the presence of echo signals and eventually to evaluate the echo parameters.

Two kinds of echoes can be found in telecommunication systems:

- Hybrid

  Generated by reflections of the incident signal at the connection between the digital unidirectional and the analog bidirectional wires of the network due to the non-ideal behavior of the hybrid devices

- Acoustic

The effect of coupling between the loudspeaker and the microphone of the telephone

A telecommunication network can be well described by a linear system; as a consequence, the echo path can be modeled as a linear filter with impulsive response $h[\cdot]$. Thus, if $x[\cdot]$ is the incident speech, the corresponding echo $d[\cdot]$ can be expressed as the convolution between $x[\cdot]$ and $h[\cdot]$.

To characterize echo signals, international recommendations suggest estimating the delay of the echo signal and echo path loss, defined as the integral of the impulse response of the echo path. [1]

A covariance analysis is initially performed on the signals acquired in the two directions of the 4-wire loop. This is to detect the presence of the echo and to estimate the echo delay.

Once the echo delay is estimated, the echo path impulse response is obtained using an adaptive filtering technique. An adaptive filter is used, which adjusts its coefficients to minimize an opportunely defined error function. This way, when the filter has converged, its output is a replica of the echo signal and its coefficients estimate the impulsive response $h[\cdot]$ of the echo path. From the estimated impulse response $h[\cdot]$, the echo path loss is easily determined.[1]

Successively, the speech signal is further processed to estimate the active speech power level and the noise power level. To this aim, a second Bayesian classifier was designed that segments the observed signal in speech intervals and noise intervals. Thus, the active speech power level is estimated from the speech periods, and the noise power level is obtained from the noise periods.

An instance of the previous algorithms should be run concurrently for each analyzed communication channel. For this reason, to optimize the performances provided by the DSP unit, a real-time multitasking kernel of an operating system has been implemented to coordinate the various operations performed by the DSP. Very satisfactory results have been obtained by using a slightly modified version of the μC/OS kernel described by Labrosse.[2]

In the DSP unit, the I/O procedures and the measurement algorithms are organized in three tasks

- TASK0
- TASK1
- TASK2

TASK0 controls the I/O operations and prepares the data buffers that must be processed for the estimation of the parameters of interest.

TASK1 performs the active speech level and the noise level measurements together with the voice-data discrimination.

TASK2 performs the echo measurements.

The three tasks are stored in the external 512 Kbyte block of static RAM. The percentile of static RAM and the percentile of available time employed by each task are reported, respectively, in the first and in the second column of Table 1.

Table 1. Percentile of Available Memory and Available Time Employed by the Task

	Available Memory (%)	Available Time (%)
TASK0	8	5
TASK1	35	20
TASK2	10	40

When the CTRL block detects an active channel in the analyzed data-stream, it starts the measurement by sending the START command to the DSP unit. To do this, it sets a predefined location of the dual-port memory, which causes TASK0 to be activated. When the data buffers are ready to be processed, TASK1 and TASK2 perform the measurements of the parameters of interest. Then the obtained values are written in the dual-port memory, where the CTRL unit collects them to be transmitted by the modem to the host computer.

Experimental Results

Many experiments have been performed to characterize the performances of the proposed instrument, which has created a wide database of tested sequences.

To estimate the probability of error of the voice-data discriminator, a number of data communications have been acquired in such a way to account for all the commonly adopted modulation techniques. A number of conversations between different speakers have been recorded in a clean environment and then corrupted with real-life disturbances.

In particular, broadband noise at different levels and echoes at different levels and delays have been added to each conversation. Then, the number of decision errors obtained by the proposed technique on the quoted data base has been evaluated. It has been seen that, on 380 voice sequences and 100 data sequences, the voice-data discrimination has always been performed correctly. The voice communications of the database have then been used to characterize the accuracy of the instrument in the evaluation of the parameters of interest.

Table 2 reports the mean and standard deviation of the estimation errors for the parameters of interest. It can be seen that the estimates are almost unbiased for all of the parameters of interest, and very low standard deviations are obtained for the estimates of the noise level and the echo delay.

Table 2. Mean and Standard Deviation of the Estimation Errors for the Parameters of Interest

	Mean	**Standard Deviation**
Act. Speech Level	0.1 [dB]	0.8 [dB]
Noise Level	0.0 [dB]	0.1[dB]
Echo Path Loss	-0.6 [dB]	2.0 [dB]
Echo Delay	0.0 [ms]	0.3 [ms]

On the other hand, higher values of the standard deviations of the estimation error are obtained for the active speech level and the echo loss, because of the intrinsic highly variable nature of speech.

Furthermore, many in-the-field experiments have been performed. In fact, the proposed instrument is currently being used to characterize the degree of quality provided by the Italian public switched telephone network.

Conclusions

A system for a in-service, non-intrusive measurement device was described using the TI TMS320C31 DSP. A simplified model of a telecommunication system was provided. Then, the hardware architecture of the instrument was summarized, the adopted estimation algorithms and the operating system kernel were briefly described. Finally, simulation results that show the accuracy of the instrument were shown and discussed.

The in-service, non-intrusive measurement device described in this application report is currently in used for the characterization of the degree of quality provided by the Italian public switched telephone network.

References

[1] ITU-T Draft Recommendation P.561, *In-Service, Non-Intrusive Measurement Device - Voice Service Measurement*, International Telecommunication Union, Geneva, September 1995
[2] J.J. Labrosse, *µC/OS. The Real-Time Kernel*, R&D Publications Inc., 1992