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Review on Implementation of Digital Music Equalization (Echo & Reverberation) Model Using Simulink and TMS320C6713 DSK

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ABSTRACT

Audio equalization is a technique which consists of boosting or cutting certain frequency components of a given signal for sound quality enhancement. Equalizer is an electronic device or type of software that increases and decreases the power of sound waves. The paper deals with the analysis of audio signals to develop insight on its frequency bands and auditory perception. This paper shows the implementation of the echo and reverberation effects using the TI's C6713 DSK. The effects are simulated using SIMULINK (The Mathworks, Inc.). Those simulation models are used to generate the DSP code for the real-time implementation. Once any of the models is complete and built, it automatically opens Code Composer Studio and compiles the Simulink block sets model. This gives us added advantage of easily writing codes on MATLAB and implementing it on the DSP processors. Echo and Reverberation are two of the simplest applications of Digital Signal Processing. DSP allows artificial echoes and reverberation to be added during mix down to simulate various ideal listening environments.

Keywords:- Code Composer Studio, Echo, Reverberation, Simulink, TI's C6713 DSK.

I. INTRODUCTION

Digital signal processing is the core technology, in rapidly growing application areas, like wireless communications, audio and increased popularity because of the various advantages like reprogramability in the field, cost-effectiveness, speed, energy efficiency etc. The key factor behind this success is the development of low-cost software. The DSP processors have gained software and hardware support [3].

This paper shows how models of Echo and Reverberation can be designed using MATLAB and Simulink and run them in real-time on the Texas Instruments C6000 DSPs family processor C6713.

Simple echo resembles FIR response whereas reverberation like IIR response which implies that echo does not implement feedback structures whereas reverberation requires feedback. Thus the frequency response of the echo system results in a comb filter. Notion behind the implementation of Reverberation is direct sound arrival is followed by reflections from all over room surfaces. Overlapping reflections are observed as reverberation. To implement reverberation, re-circulating delay lines are employed to create an artificial impulse response which gives less computation and storage, but complex to get satisfactory audio quality [1].

Music equalizers are either devices or software which are used for amplifying and/or attenuating predetermined frequency bands. In many areas of sound processing like audio recording studios, or voice/music signals transmission to the audience during a concert, equalizers are assembled for sound control and enhancement [2].

Equalizers can be classified in two main categories:

1. Graphic equalizers
2. Parametric equalizers

Parametric equalizers can adjust parameters of the filters, whereas graphic equalizers can adjust the relative positions of the sliders to build a graphic picture of the desired magnitude response. For graphic equalizers, only boost or cut can be controlled with sliders by keeping the centre or middle frequency unchanged. A graphic equalizer comprises a bunch of band-pass filters each with fixed centre frequency and a predefined bandwidth, where the first and last bands are respectively low-/high pass filters [2].

The main objective of this paper is to design models of Echo and Reverberation for the Texas Instruments to run the model in real-time on the Texas Instruments C6713 DSK.

The remainder of this paper is organized as follows. In section II, it will introduce the notion of echo & reverberation. Section III will define the hardware & software requirements. In section IV, it will elaborate proposed system for implementing the equalizer on DSP board with the Simulink model. Section V describes the procedure to execute the Simulink model in real-time using a TMS320C6713 DSP board. In Section VI, conclusion is mentioned.

II. ECHO & REVERBERATION

A. Echo

In audio signal processing, an echo is a reflection of sound, approaching at the listener later than the direct sound. A true echo is a single reflection of the original sound. The time delay is the ratio of the extra distance to the speed of sound. An echo can be realized as a signal wave that has been reflected by a medium discontinuity in the propagation medium and returns with sufficient magnitude and delay to be perceived by human ear. So that in echo effect, the true sound and the artificial sound are clearly separated, so that human hearings can tell the difference.

Echo is audible because the speed of sound is relatively slow, about 3 43 meters per second. If we consider only one echo path, then an echo can be simulated using the following equation:

Output = Input + Delayed Input × Gain

Where Gain < 1, due to losses in the echo path [1].

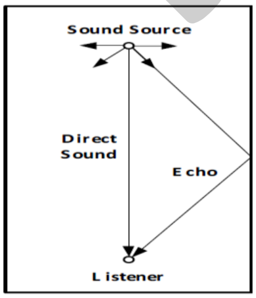

Fig. 1 Basic principle of Echo

As seen in Fig. 1, the signal propagates from the source to the listener in two paths. First, the signal from the source goes directly to the listener. Second, the signal goes to the wall and then reflected to the user. The second process will take more time than the first process, so the listener will hear two sounds in a different period of time. The signal power from the second process will be attenuated due to the reflection process.

When dealing with audible frequencies, the human ear cannot recognize the identity of an echo from the original sound if the delay is less than 1/10 of a second. Thus, since the velocity of sound is approximately 343 m/s at a normal room temperature of about 20°C, the reflecting object must be more than 16.2 m from the sound source, for an echo to be heard by a person at the source. In most situations with human hearing, echoes are about one- half second or about half this distance, since sounds grow fainter with distance. The strength of an echo is frequently measured in dB sound pressure level SPL relative to the directly transmitted wave [6].

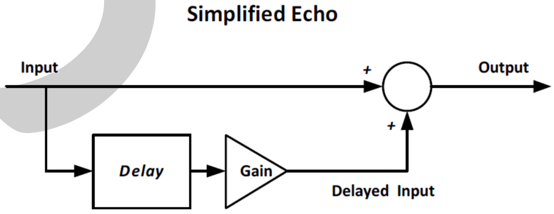

Fig 2. Basic system representing echo over a single path

B. Reverberation:

Reverberation is the persistence of sound in a particular space after the original sound is removed. A reverberation, or reverb, is observed when a sound is created in an enclosed space causing multiple echoes to build up and then slowly decay as the sound is damped by the surrounding walls and air. It is the sum of all sound reflections. This is most noticeable when the sound source stops but the reflections continue, decreasing in amplitude, until they can no longer be heard.

The principle of reverberations is more like echo, but in reverb the sound reflections comes very often in a short period of time.

In comparison to a distinct echo that is 50 to 100ms after the initial sound, reverberation is many thousands of echoes that arrive in very quick succession i.e., 0.01 – 1 ms between echoes. With the elapse of time, the intensity of the multiple echoes is reduced till the echoes are inaudible [1].

In reverberation, the output is derived from both the input and the previous output:

Output = Input + Delayed Output × Gain

There are two important parameters in reverberation, which are:

- Predelay, is the period amount of time of the first sound reflection
- Reverb decay, is the period amount of time of reverb since the input stops.

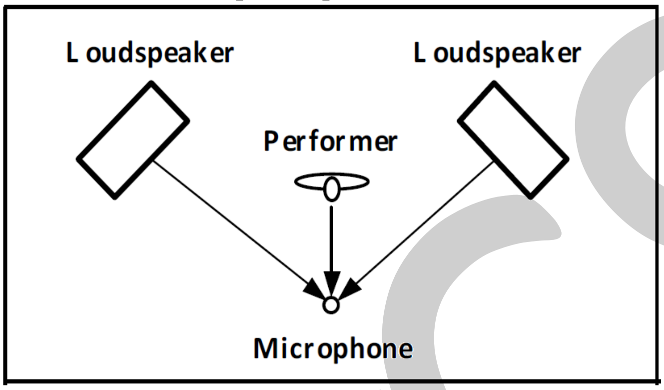

Fig. 3. Basic principle of reverberation

Fig. 3 shows the mechanism of reverb effect. Every reflection from the source can be a new sound source. In reverb effect, usually the listener cannot differentiate between the original sound and the reverb sound. The length of this sound decay, or reverberation time is the term of special consideration in the architectural design of large chambers, which need to have specific reverberation times to achieve optimum performance for their intended activity.

Digital reverberations use various signal processing algorithms in order to create the reverb effect. Since reverberation is essentially caused by a very large number of echoes, simple reverberation algorithms uses multiple feedback delay circuits to create a large, decaying series of echoes [6].

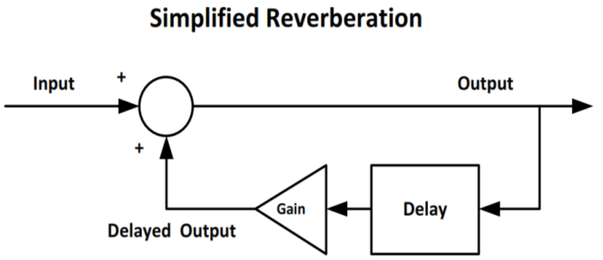

Fig. 4. Basic principle representing reverberation over a single path

III. SYSTEM REQUIREMENT

In software part, the MATLAB package consists of a high-level technical computing language and an interactive environment for algorithm development, data visualization, data analysis and numeric computation. Within the Matlab environment, Simulink is a Matlab toolbox that differs from the other toolboxes, both in this special interface and in the special "programming technique" associated with it.

Simulink is an interactive tool for modeling, simulating and analyzing dynamic systems, including controls, signal processing, communications and other complex systems with a graphical interface specially developed for this purpose. It is advantageous in programming DSP algorithms using Matlab. This makes ease of coding, able to use a powerful set of inbuilt functions and seamless link between Matlab and Simulink.

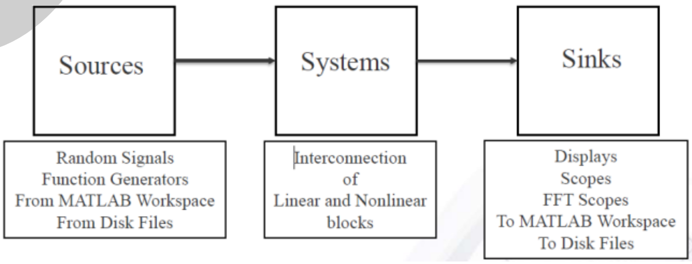

Fig 5. Typical Simulink Model

In hardware part, Digital signal processors such as the TMS320C6x (C6x) family of processors are like fast special-purpose microprocessors with a specialized type of architecture and an instruction set appropriate for signal processing. The TMS320C6x are the first processors to use velociTI architecture, having implemented the very-long-instruction-word (VLIW) architecture which makes it most powerful processor. That in turn makes it most suitable option for numerically intensive calculations. The C6x notation is used to designate a member of Texas Instruments' (TI) TMS320C6000 family of digital signal processors. The TMS320C67x is a floating point processor, with 32-bit integer support. The C6713 DSK is a low-cost standalone development platform that enables users to evaluate and develop applications for the TI C67xx DSP family [1].

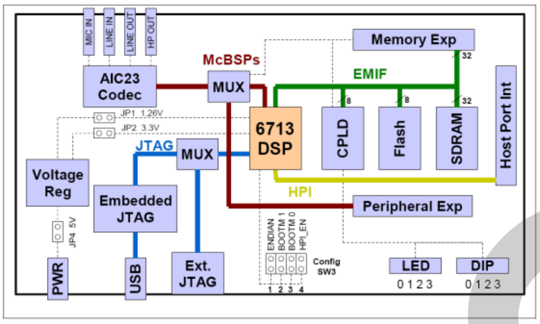

Fig. 6. Block diagram of TMS320C6713 DSK

DSK TMS320 C6713 is a DSP board, a platform where someone can build a signal processing application by writing a program in the DSP board. How the DSP processes signal depends on how the user program the DSP board. The output signal will be release through line out/headphone.

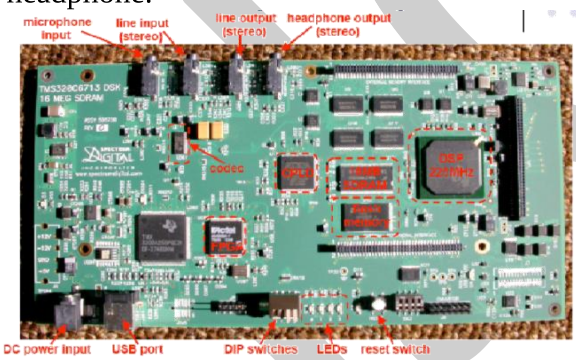

Fig. 7. Block diagram of TMS320C6713 DSK

The DSP on the 6713 DSK interfaces to on-board peripherals through a 32-bit wide EMIF (External Memory InterFace). The DSP interfaces to analog audio signals through an on-board AIC23 codec and four 3.5 mm audio jacks (microphone input, line input, line output, and headphone output). The codec can select the microphone or the line input as the active input. The analog output is driven to both the line out (fixed gain) and headphone (adjustable gain) connectors.

The DSK includes 4 LEDs and a 4 position DIP switch as a simple way to provide the user with interactive feedback.

An included 5V external power supply is used to power the board. On-board switching voltage regulators provide the +1.26V DSP core voltage and +3.3V I/O supplies. The board is held in reset until these supplies are within operating specifications.

The DSK comes with a full complement of on-board devices that suit a wide variety of application environments. Key features include [4]:

- A Texas Instruments TMS320C6713 DSP operating at 225 MHz.
- An AIC23 stereo codec
- 16 Mbytes of synchronous DRAM
- 512 Kbytes of non-volatile Flash memory (256 Kbytes usable in default configuration)
- JTAG emulation through on-board JTAG emulator with USB host interface or external emulator

To develop an application with DSK, Texas Instrument provides software called Code Composer Studio (CCS). Code Composer Studio™ (CCStudio) is an integrated development environment (IDE) for Texas Instruments (TI) embedded processor families. CCStudio comprises a suite of tools used to develop and debug embedded applications. It includes compilers for each of TI's device families, source code editor, project build environment, debugger, profiler, simulators, real-time operating system and many other features. The intuitive IDE provides a single user interface taking you through each step of the application development flow. Familiar tools and interfaces allow users to get started faster than ever before and add functionality to their application [5].

Code Composer communicates with the DSK through an embedded JTAG emulator with a USB host interface. The DSK can also be used with an external emulator through the external JTAG connector [2].

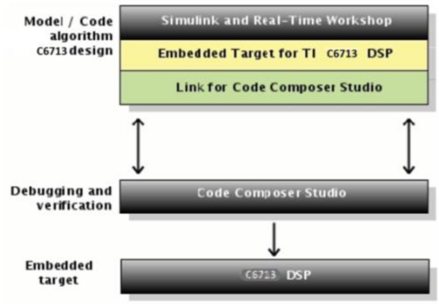

Fig 8. Flow stages of project

IV. SYSTEM DESIGN

The Echo and Reverberation systems are designed and simulated using Simulink. The basic principle behind the design is the delay. The design principle of Reverberation is same as Echo with a small change which is the feedback. Reverberation can be designed using feedback and without feedback. Each effect is selected using the DIP switch in the DSK [2] [3].

The system in Fig. 9 is a Music Equalizer that particularly alters the energy levels of the audio data in one or more different frequency bands in order to change the characteristics of the audio data. This equalizer facilitates boosting (or suppression/attenuation) of frequencies between the source of a sound and the output of the sound.

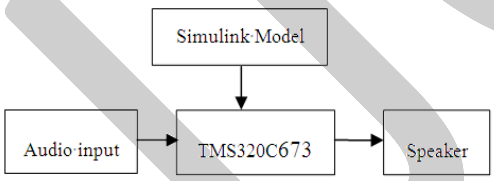

Fig 9. System implementing music equalizer using TMS320C6713

The equalizer system of Fig. 9 has two major parts, the DSP Processor and the Simulink model.

V. EXECUTING SIMULINK MODEL ON DSK

Steps to follow in building and executing the Simulink model on C6713 DSK [8]:

1. Build the Simulink model using the blocksets from Simulink Library Browser in Matlab.
2. Open “TIC6000 Target Preferences” from Simulink Library Embedded target Content for TI C6000.
3. Select 6713DSK from the list of DSK’s available and add C6713 DSK board support components to the model (like ADC, DAC, RESET, DIP Switch and LED).
4. Run the CCS software in background.
5. Apply the input music signal to the Line In input of DSK and connect the speakers to the Line Out.
6. Open the Simulink model of equalization and press CNTRL + B to build an equivalent ‘C’ language code in CCS. Simulink starts communicating with CCS. The CCS automatically gets open and the .out File will be loaded into the DSP.
7. Select the band by setting the DIP switch and the corresponding equalized sound will be heard through the speaker.
8. To stop model execution, click the Reset DSK block or use the Halt option in CCS.

VI. CONCLUSION

The design of digital audio effects can be applied to the DSP board. The magnitude of the outputs may vary with respect to Delay and Gain (decay factor). When gain is increased the magnitude increases which will be highly significant in Reverberation where the system becomes unstable at Gain above 1. If delay is reduced keeping the Gain constant the magnitude also will reduce. These digital effects are used in real time applications. The input signal of audio effect can be modified by using a music instrument or voice as input.

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