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Speech Endpoint Detection Using Finite State Machine Builder

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Abstract*:* IIn this paper, we use DSP Builder to realize the speech signal endpoint detection based on short-time energy variation. In the design, we use the LUT (look-up table) window design and FSM (finite state machine) to simplify the hardware circuit and improve the operating speed. Using DSP Builder to preprocess the speech signal (pre-emphasis, framing and windowing),to calculate the signal short-time energy of per frame, to convert the energy to state value according to the result of speech energy of each frame compare with the energy threshold, then determined the endpoint of speech signal by energy threshold and energy logic state numeric sequence. The results of simulation show that the method can detect the beginning point and ending point of speech signal effectively.

Keywords Endpoint detection; DSP builder; Finite state machine; Energy changing

INTRODUCTION

Speech endpoint detection is to identify non-speech signal and speech signal in noise background environment, and determine the beginning point and ending point. That is considered as one of the key preprocessing components in automatic speech recognition. [Zhao Lihua *et al.*, 2010] Recently, there are many endpoint detection methods, such as short-time energy, zero-crossing rate, spectral analysis, spectral entropy, wavelet transform and hidden Markov model (HMM) and so on, many methods are based on software and the calculation is large and complex and therefore can’t be performed in real-time detection.[Zhang Dexiang *et al.*, 2010] But with DSP Builder speech endpoint detection algorithm greatly improves the speed of computation, especially in embedded systems, satisfies hardware environment and the real-time requirements. Therefore, this paper mainly uses the DSP Builder tool to preprocess the input speech signal (pre-emphasis, framing, windows) and using short-time energy speech endpoint detection algorithm. The results show that this method has some validity and feasibility.

SPEECH ENDPIONT DETECTION USING SHORT-TIME ENERGY

Basic theory of short-time energy to detect the endpoint of speech signal

For noisy speech signal, the energy of the noisy speech segment is sum of the noisy signal energy and speech signal energy [Shen Hongli *et al.*, 2008]. Thus, the energy of the speech segment is larger than the energy of noise segment. At high SNR conditions, by calculating the short-time average energy of the noisy speech signal to distinguish speech segments from background noise. Energy of per frame signal obtained by the following formula:

$$E_j = \sum_{i=0}^{N-1} x^2(i) \tag{1}$$

Where E_j is the energy of the j-th frame, x_i is the input speech signal, N is the frame length.For high SNR speech signal, noise energy En is small, and the speech signal energy En is significantly high, which can be used to distinguish the beginning and ending points of the speech signal.

Basic theory of short-time energy to detect the endpoint of speech signal using DSP builder

According to short-time energy during endpoint detection and possible situation of duration, the whole endpoint detection process is divided into 6 states: initial state (S0), the silence state (S1), rising energy state (S2), energy continuous state (S3), decreased energy state (S4) and the energy rise-fall state (S5). Changes depend on its state transition condition (G1 is rising state energy threshold, G2 is continued state energy threshold, E is the current frame energy, Ci is the number of frames in the state i, (1) (2) (3) are priority of the state transition to the next state). Energy state change is shown in Figure 1.

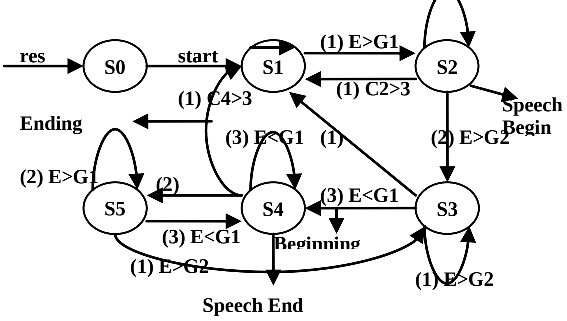


Figure1. Diagram of the energy state change

After the system reset, using the starting 16 frames to calculate the average energy of the noise signal and set it as threshold. Computing the average energy of each frame, obtain the average energy E_{ave} of 16 frames and the maximum frame energy $E_{\max}$,according to $G_i = (E_{\max} - E_{ave}) \times a_i$,i=1.2 two dynamical threshold values, where a1 = 3, a2 = 6. G1is starting energy threshold, G2 is sustained energy threshold.

When the start signal is active, enter the S1 state, if one frame energy of speech exceeds the initial energy threshold G1, then enter the S2 state, otherwise stay in the S1 state.

When entering the S2 state, to judge whether more than 3 frames in S2 state.

1) If there are more than 3 frames and not enter S3 state, give up the beginning point, that is the pulse.
2) If a frame energy exceeds the threshold value G2 then enter S3 state, if maintained for more than 10 frames in the S3 state, it is the beginning point of speech, then enter the S2 state, and its first frame is the start frame, otherwise considered invalid noise, back to S1 state.

After judging the beginning point, then continue to calculate the speech energy, there are two conditions:

3) When the speech energy decreased below G1 then enter S4 state, if the number of frames in S4 state more than three, we think the speech is over. And the entering frame of S4 state is the endpoint of the speech.
4) If entering S4 state, speech energy is quickly increased higher than G1, then enter the S5 state. In S5 state if the energy is higher than G2 then enters S3 state, if the energy less than G1 then back to S4 state, which is to prevent a small pause in speech judged as speech over.

Effective speech is from the S2 state's first frame to the S4 state's first frame, assuming that the minimum effective length is 15 frames, that is the sum of rising state's frames and continuing state's frames must be more than or equal to 15.

ENDPIONT DETECTION USING DSP BUILDER

Pre-emphasis of speech signal

Diagram of pre-emphasis model is shown in Figure 2, Delay module is a delay of 1 clock cycle, Barrel Shifter module is a shift register, right shift 4 bit (equal to divide by 24).

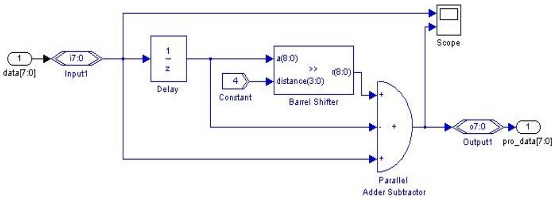


Figure2. Diagram of pre-emphasis model

Sub-frame process of speech signal

Diagram of sub-frame process model is shown in Figure 3.

In Figure 3, the input of pre_data is the data after pre-emphasis and the output of enframe_data is the data after sub-frame.FIFO1 is used for storage data, to control data can be read and written. FIFO2 is variable speed register which can read and output. DMUX is data selector, using sel to control the output data. Signalin is subsystem (shown in Figure 4) which contains mod 256 Counter module and Increment Decrement module.

The process of register data: Signalin subsystem's count value is 128 then generate FIFO1 read enable signal, When Counter value is 0 ~127 then reading FIFO1, and writes the read out data to FIFO2, DMUX output FIFO1 data. When Counter value is 128~255 read FIFO2, DMUX output FIFO2 data. After pre-emphasis speech data can be continuously written to FIFO1, real-time dynamic sub-frame speech.

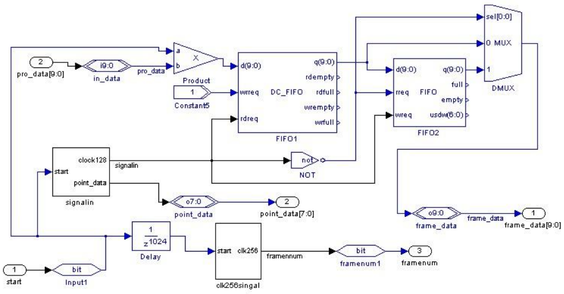


Figure3. Diagram of sub-frame model

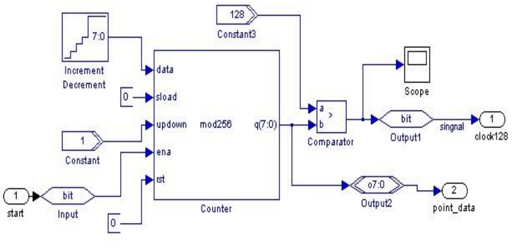


Figure4. Diagram of Signalin subsystem model

Windowing the speech signal and calculating energy of per frame

Diagram of windowing and calculating energy of per frame is shown in Figure 5.

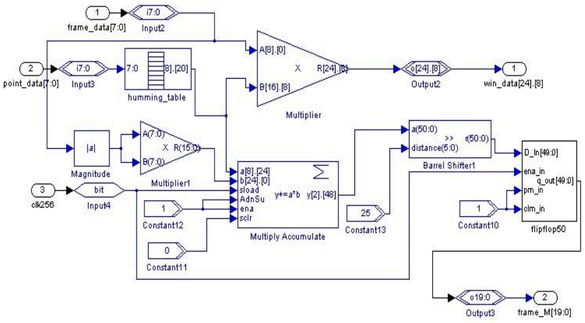


Figure5. Diagram of windowing and calculating energy of per frame

We use LUT (look up table) module to realize Hamming window. The LUT input formula is $0.54-0.46*\cos([0:2*pi/(2^\wedge 8-1):2*pi])$, where n is between 0 and 255.To process the sub-frame and window at the same time, the address is the value of mod 256 counter of sub-frame.

Recorded speech signal frames

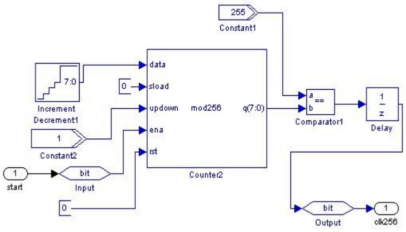


Figure6. Diagram of subsystem model clk256signal

In Figure 6 is Clk256signal subsystem which is to produce 256 points periodic pulse signal. Since the speech data is delayed after the sub-frame process, the delay is calculated as about 4 frames that are 1024 points (clock cycles), so join a 1024-point delay in the sub-frame model to synchronizes clk256signal and output of speech data. When clk256signal count 256 points, and then produce a high level, in order to control the recording speech signal frames model.

Diagram of recording speech signal frames is shown in Figure 7.

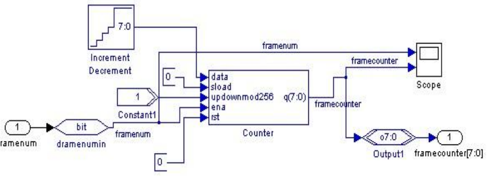


Figure7. Diagram of recording speech signal frames

Counter module enable input signal is output of clk256signal subsystem, counter module is used to record the number of frames, framecounter is output and that is frame number.

Setting dual-threshold

Diagram of setting dual-threshold is shown in Figure 8.

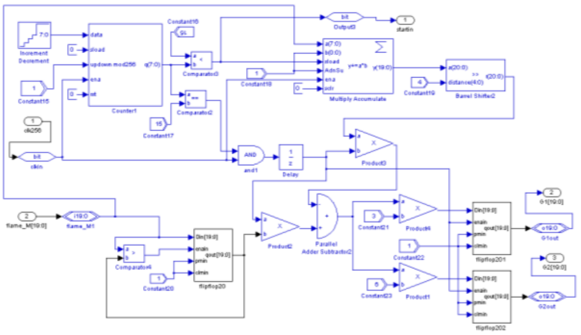


Figure8. Diagram of setting dual-threshold signal frames

The processes of design are as follow:

Firstly, calculate the average energy: calculate the total energy of the first 16 frames, with a multiply accumulator Multiply Accumulate module to calculate energy of 16 frames, and then divided by 16 to obtain the average energy (Barrel Shifter shift register to the right by 4 bit).

Secondly, find the maximum energy: using comparator Comparator4 module to compares the energy of current frame and the energy of last frame, the comparison output is 1 or 0, and that is enable signal of 20 bit D flip-flop. Here En is the current frame energy, Eb is the last frame energy, when En> Eb, the output of D flip-flop is En, set En as the input of Comparator4 module in order to continue compare, when En< Eb, D flip-flops keep energy value of last frame, set Eb as the input of Comparator4 module in order to continue compare.

Finally, dual-threshold setting: according to $G_i = (E_{\max} - E_{ave}) \times a_i$, i=1, 2 two dynamical threshold values, where a1 = 3, a2 = 6. G_1 is starting energy threshold, G_2 is sustained energy threshold. First calculating the difference between the $E_{\max}$ and E_{ave} by Parallel Adder Subtractior module, and then obtain G_1 and G_2 by Product module. With D flip-flop to maintain G_1, G_2 signal.

FSM design

According to the energy changes and dual-threshold, design FSM with 6 states transition, diagram of FSM model is shown in Figure 9.

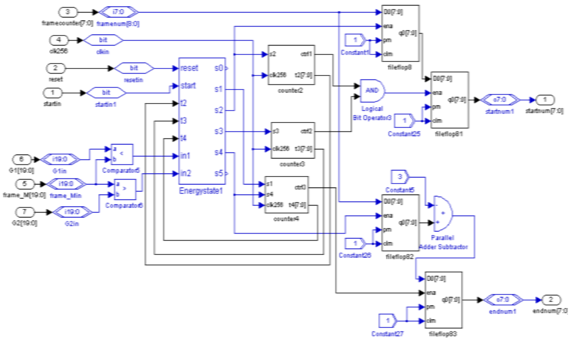


Figure9. Diagram of FSM and endpoint detection

In Figure 9, the state machine Energystate1 is State Machine Table of State Machine Functions Library, which can set states and state transition conditions. Reset is a system reset signal, start is starting endpoint detection signal, G1 and G2 are threshold values, the input frame-M is energy of each frame, counter2, counter3 and counter4 are subsystem models, respectively contains a counter to count frame number in state S2,S3andS4, startnum is frame number of beginning point and endnum is frame number of ending point. Where D flip-flop filpflop8, filpflop81, filpflop82, filpflop83 are 8 bit.

The whole endpoint detection system

The subsystem is integrated into a whole system as shown in Figure 10.

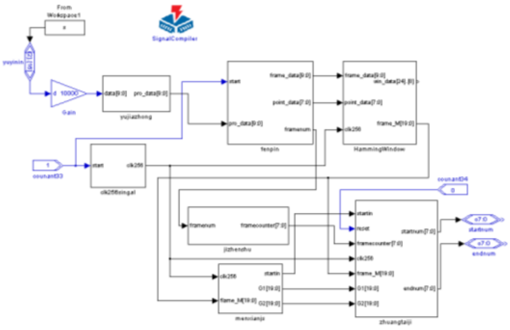
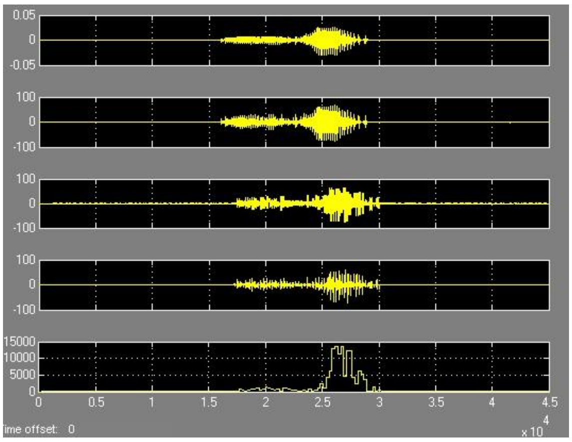


Figure10. Diagram of the whole system

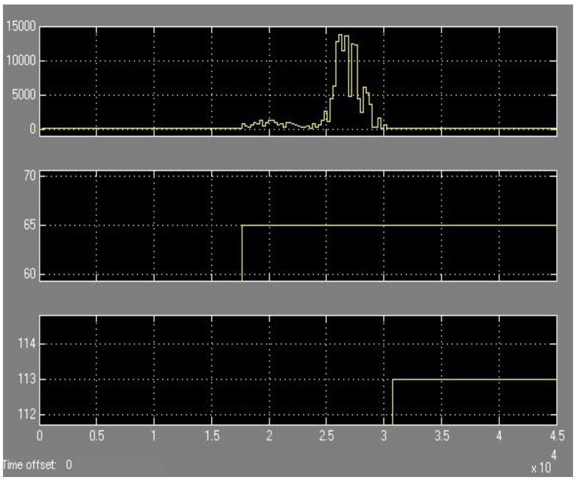
EXPERIMENT AND RESULTS

The results of endpoint detection using DSP builder

All of the experimental results are compiled in the DSP Builder software. The speech signal is collected by computer sound card which is wav format. The speech samples using 8 kHz sampling frequency and 8bit quantification, 16 bit sampling precision. The content of the speech signal is Chinese word “1” and “2”. We test the effect of endpoint detection.



(a)



(b)

Figure11 Result of the endpoint detection using DSP Builder

In order to obtain the result of speech signal process, we add scope model to the DSP Builder to observe the wave of every process. The result is shown in Figure 11.

In Figure (a) is original speech signal wave of Chinese word “one, two”, and waves of pre-emphasis, sub-frame, windowed, each frame energy. In Figure (b) is diagram of the energy of each frame and beginning point and ending point. From the Figure (b) we can find the beginning frame number is 65 and the ending frame number is 113.It can detect the endpoint of speech signal clearly.

Compared with the result of MATLAB software

In order to show the effectiveness of endpoint detection using DSP Builder, we compared with the result of endpoint detection using MATLAB software.

Figure 12 is the result of the endpoint detection with short-time energy using MATLAB software. From the figure we can see the beginning point frame is 63 and the ending point frame is 112.

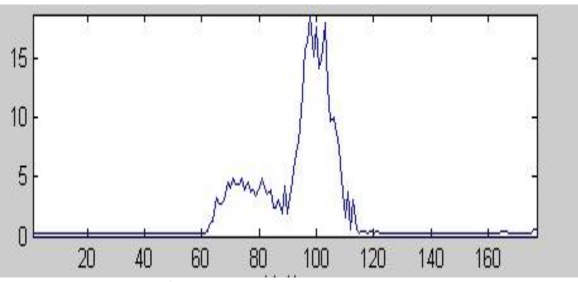


Figure12 Result of the endpoint detection with short-time energy using MATLAB software

From the simulation results of endpoint detection using DSP Builder and MATLAB software, we can find that the results DSP Builder endpoint detection is accurately. The DSP Builder model can change into hardware language then programming on FPGA or CPLD. Therefore, this design has a strong practical using.

CONCLUSION

This paper use DSP Builder to realize endpoint detection of speech signal. The results of simulation show the feasibility of endpoint detection using DSP Builder. FSM design the computational complexity and improve the computing speed, moreover, reduces the development time greatly, DSP Builder graphical modeling have advantages which are flexible design, easy to modify and the design process clearly. The ultimate goal of DSP Builder is designed for real-time processing on the hardware chip. Thus this design has good practicability.

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