DSP LSI을 이용한 DTMF 수신기의 구현에 관한 연구

On Implementing the Digital DTMF Receiver using DSP LSI

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요 약

DSP LSI을 이용하여 디지탈 DTMF 수신기를 구현하는 방법으로는 IIR 디지탈 필터, FIR 디지탈 필터, Counter 방법, DFT방법, FFT 방법 및 PARCOR 방법 등이 제안되어 왔다. 그 중에서도 IIR 디지탈 필터를 이용한 방법은 기존의 아나로그 DTME 수신기를 그대로 디지탈화 한 것이기 때문에 성능이 제일 우수한 것으로 알려져 있다. 그러나 IIR 디지탈 필터를 이용하여 그것을 구현할 때 필터의 계수, roundoff 잡음, overflow 등 고려해야 할 사항이 많다. 본 논문에서는 이러한 문제점들을 해결하면서 CCITT 사양들을 만족하는 디지탈 DTMF 수신기 구현에 관한 연구결과를 제시하였다[1] DSP LSI 을 이용해서 수신기를 hardware 제작할 때 이 결과들을 수정없이 이용할 수 있다고 기대된다.

ABSTRACT

The following methods are proposed for implementing digital dual tone multifrequency (DTMF) receiver using general purpose digital signal processor (DSP) large scale integration (LSI): using infinite impulse response (IIR) digital filters, finite impulse response (FIR) digital filters, period-counting algoithm, discrete fourier transform (DFT), fast fourier transform (FFT), or partial correlation (PARCOR) analysis. The method using IIR digital filters is known to be the best of these since it is just the digital version of the conventional analog DTMF receiver. But, in implementing it using IIR digital filters with available general purpose fixed-point DSP LSI, there are several problems: filter coefficient, roundoff noise, overflow, and so on. This paper describes results of studies on an all-digital DTMF receiver overcomed for these problems and meeting the requirements of CCITT recommendations [1]. So it is expected that these results can be used for the hardware realization of the receiver with available DSP LSI without any modification.

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I. INTRODUCTION

The convenience of dialing a phone with pushbuttons instead of a rotary mechanism plays a key role in spreading the popularity of pushbutton service. Pushbutton service is a voice (or in-band) frequency signalling system in which any one of 16 digits may be transmitted by simultaneously sending two tones. The frequency of one of the tones may be either 697, 770, 852, or 941 Hz (called the low frequency group) and the frequency of the other tone 1209, 1336, 1447, or 1633 Hz (called the high frequency group). The dual tone multifrequency (DTMF) receiver receives and decodes the DTMF tones. The receiver must tolerate frequency shifts in the transmitter, operate over a wide dunamic range in the presence of noise, and be insensitive to speech digit simulation).

Digital devices superiority over analog devices has been recognized with advances in large scale integration (LSI) and digital signal processing technologies. Digitalization has the effect of making a device small and of simple manufacture and maintenance. Especially for dual tone multifrequency (DTMF) receiver which must satisfy severe specifications, it is attractive that aging degradations become negligible when it is digitalized.

Various schemes for the digital DTMF receiver are shown in Table 1. Among these schemes, the 11R digital filter scheme is hopeful in that band rejection filters are obtainable by this scheme. It is expected that erroneous operation by user voices will be prevented by

Table 1. Various Schemes for a Digital DTMF Receiver

Schemes	General Descriptions
IIR Digital Filters (IIR DF)	Filters in analog receiver are replaced by IIR digital filters. Same characteristic as for the analog receiver can be expected in this scheme.
FIR Digital Filters (FIR DF)	The principle is the same as that of IIR DF except for being replaced by FIR digital filters instead of IIR digital filters.
Discrete Fourier Transform (DFT)	Input signal is transformed into fourier series. Fourier series frequencies are pushbutton signaling frequencies
Fast Fourier Transform (FFT)	The principle is the same as that of DFT. FFT detection frequencies do not coincide with the pushbutton signaling frequencies. So, the FFT scale becomes large.
Period-Counting Algorithm (CNT)	After input signals are separated into two groups using digital filters, the limiter output signal frequency is counted by a clock. This scheme requires another facility to protect erroneous operation caused by voice.
Partial Correlation (PARCOR)	This scheme detects pushbutton signaling frequencies based on computed PARCOR coefficients.

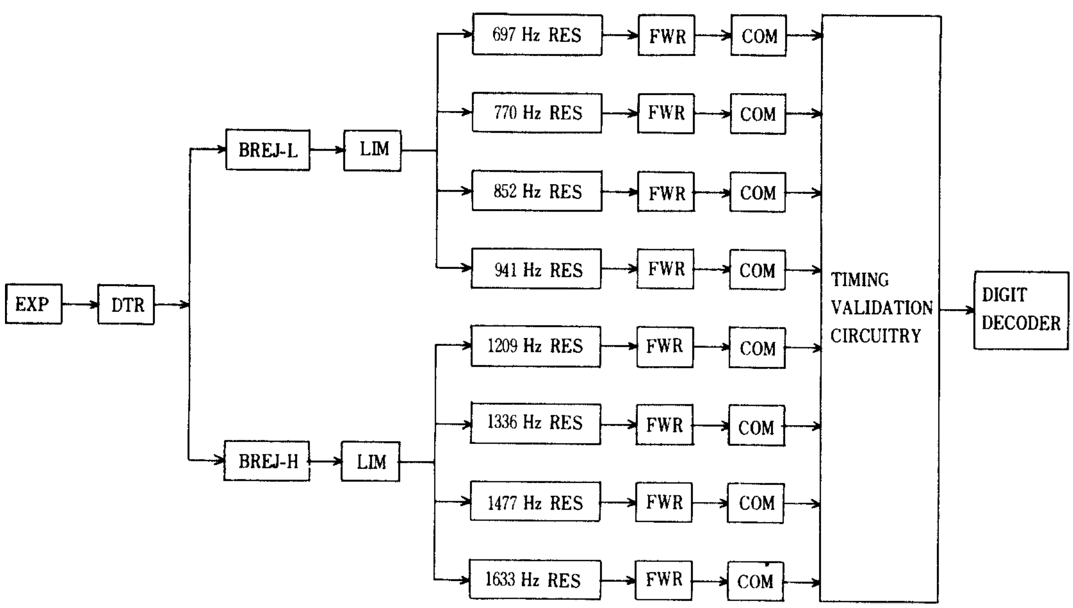
band rejection filters together with limiters, as is the case of a conventional analog DTMF receiver[5].

In the following, this paper describes results of studies on an all-digital DTMF receiver for pushbutton service using IIR digital filters optimized with regard to the problem of filter coefficient, roundoff noise, overflow and so on.

Various general purpose digital signal processor (DSP) large scale integration (LSI) are on the market and we can easily implement the digital signal processing algorithms with them. Yet they allow only fixed-point arithmetic, and so we encounter many problems: filter coefficients, roundoff noise, overflow, and so on, when we implement digital filters with available general purpose fixed-point DSP LSI.

Several analysis of the roundoff noise output from a digital filter with fixed dynamic range (i.e., implemented using fixed point arithmetic) have appeared in the literature [2],[3]. Comparisons between the roundoff noise outputs of different circuit configurations for a digital filter have been particular interest because of desire to maximize some measure of the output signal-to-noise ratio.

The realization of digital filters by cascading second-order sections has many desirable features, such as better noise performance than the direct realization and permitting a modular realization of high-order digital filter in a flexible manner.



EXP : Expander
DTR : Dial Tone Rejection Filter (fourth-order elliptic)
BREJ-L : High Frequency Group Band Rejection Filter (sixth-order elliptic)
BREJ-H : Low Frequency Group Band Rejection Filter (sixth-order elliptic)
RES : Channel Resonator (second-order)
FWR : Full Wave Rectifier
COM : Comparator

Fig. 1. Difital DTMF Receiver Function Blockdiagram

When a fixed-point digital filter is realized under dynamic range constraints, the resulting roundoff error due to of finite word-length is highly dependent upon the pole-zero paring and ordering of the sections[4],[5].

The heuristic optimization method to produce an optimal pole-zero paring and ordering is applied to all the digital filters of DTMF digital receiver[4].

The receiver is particularly suited for systems that operate on signals that have been encoded into a mu-255 pulse coded modulation (PCM) format.

An incoming signal first passes through DTR (Dial-Tone Rejection) filter that filters out dial tones, ring tones, and busy tones. Next, the signal passes through two parallel band rejection filters, BREJ-H to reject the four low-group tones and BREJ-L to reject the four high-group tones. The output of each band rejection filter passes through four parallel resonators tuned at each center frequency. Each resonator is connected to a threshold detector which, in 40 ms, makes a determination of whether the tone was present or absent. The function block-diagram is shown in Fig. 1.

II. DIMF RECEIVER REQUIREMENTS

For use in PABX or central office applications, there are known requirements that a DTMF receiver mut meet. These are shown in Tables 2 and 3, and have been influenced greatly by CCITT recommendations[1].

While most of the criteria in Tables 2 and 3 can easily be met individually, it becomes more difficult to meet them collectively. In particular, proper talk-off performance (speech simulation of valid tones) is a more difficult task than one might suspect.

Table 2. System Requirements

Norminal frequencies Low frequency group High Frequency group	 697, 770, 852, 941 Hz 1209, 1336, 1477, 1633 Hz
Allowable frequency deviation from the norminal value	- 2.0 ~ 2.0%
Unallowable frequency deviation from the norminal value	less than - 2.8% or greater than 2.8%
Allowable signal level (for one tone)	- 3 - 24 dBm
Unallo wable signal value (for one tone)	less than - 29 dBm
Allowable level difference between two tones (twist) for norminal frequency for 2.0% deviated frequency	 less than 15 dB less than 5 dB
Response time	24 ~ 40 ms

III. DIMF RECEIVER DESIGN

In DTMF, one low frequency (out of four) and one high frequency (out of four) are transmitted. In addition, dial tone may be present.

The DTMF receiver function blockdiagram is shown in Fig. 1. Input signal, according to mu-255 PCM encoding law, must be expanded to the linear code used in the receiver. According to PCM encoding law, 8 bits PCM (128 levels) corresponds to 14 bits (8159 levels) linear code. EXP expands 8 bits PCM to 14 bits linear code.

DTR is a dial tone rejection filter that attenuates dial tones (350 and 440 Hz in our country).

And BREJ-L is a band rejection filter that passes the low frequency group while attenuating the high frequency group, and BREJ-H does just the reverse.

The output of each band rejection filter go into hard-limiters. The limiter has two threshold levels for each signaling frequency group, that is, +THL, -THL for the low group and +THH, -THH for the high group. The limiter input signal changes the sign of the threshold level as the input signal exceeds the present threshold level. Then, the limiter output produces a rectangular wave of 50% duty factor, whenever the input signal exceeds the threshold level. The limiter output level can be selected independently from the input signal level. It is preferable to limiter output level LOUT to be large, but smaller than the overflow level in resonators. Therefore, the inequality

$$4 \cdot \mathrm{LOUT} \cdot 10^{A/20} / 3.14 < 1 \qquad (1)$$

holds where value A is maximum resonator gain.

Each limiter output drives a set of second-order channel resonators. The filters are tuned to the signaling frequencies and have a pole-Q of 16.

The channel resonator outputs are full wave rectified and then drive detector or time-delay comparator circuits which sense a signal level that is greater than a fixed threshold VTH. The threshold is set to 2 dB below the level that would be present if a sine wave of the norminal frequency and amplitude were input to the limiter. If the signal to the detector drops below the threshold, the detector continues to indicate output for a fixed hold time. An output on the detector indicates presence of the corresponding DTMF tone.

The presence of DTMF frequencies must persist for a certain minimum interval of time and the presence of both the low frequency and the high frequency must be checked before the input signal is validicated as a DTMF addressing digit. These timing considerations are performed by the timing validation circuitry. Timing requirements are shown in Table 3.

Table 3. Timing Requirements

DTMF receiver must

1) detect proper DTMF signals if their duration is or longer.
2) not detect two pairs whose duration is 20 ms or less.
3) detect as two separate DTMF signals any separation of valid tone pairs which is 35 ms or greater.
4) not detect as two distinct DTMF signals if the separation of valid tone pairs is 5 ms or less.

The digital filters in the DTMF receiver are shown in Fig. 1. DTR, BREJ-L and BREJ-H are designed by using elliptic filters in order to flat characteristics in pass bands, and then bilinear z transformation is used to transform from s-domain to z-domain.

Each resonator is also obtained by z transformation from s-domain to z-domain. The system function derived is

$$H(z) = \frac{1-b}{2} \cdot \frac{1 - z^{-2}}{1 + a z^{-1} + z^{-2}}, \qquad (2)$$

where

$$wd = \tan(wa. \times T/2), \qquad (3)$$

$$a = \frac{2 \cdot (wd^2 - 1)}{wd^2 + wd/Q + 1}, \qquad (4)$$

$$b = \frac{wd^2 - wd/Q + 1}{wd^2 + wd/Q + 1}. \qquad (5)$$

wa is signaling angular frequency in the analog domain, T is sampling interval and g is a pole-Q of the channel resonator.

All the digital filters of the DTMF receiver are realized by cascading second-order sections since it has many desirable features, such as better noise performance than the direct realizations and permitting a modular realization of higher-order filter in a flexible manner[8].

When a fixed-point digital filter is realized under dynamic range constraints, the resulting roundoff error due to finite word-length is highly dependent upon the pole-zero pairing and ordering of second-order sections[4],[5].

S, the heuristic optimization method to produce an optimal pole-zero pairing and ordering is applied to all the digital filters of the DTMF receiver[4]. The procedure chooses a random pole-zero pairings and performsa a local optimization. This is repeated a number of times and the best of these local optima is taken as the near optimal pole-zero pairing.

The frequency characteristics for DTR, BREJ-L, BREJ-H and 697 Hz RES are shown in Fig. 2, 3, 4 and 5 respectively.

V. SIMULATION AND RESULTS

We investigated by computer simulation that the designed receiver meets the requirements of CCITT recommendations, by displaying the results of each stage on the IBM PC graphic

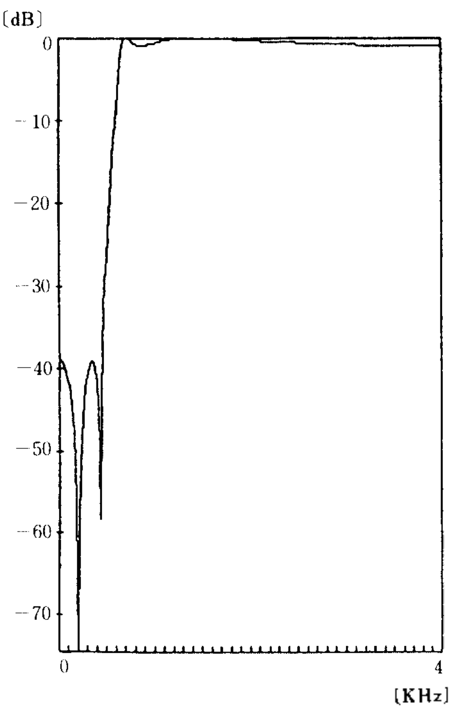


Fig. 2. Frequency Characteristic for DTR (vertical = 10 dB, horizontal = 100 Hz)

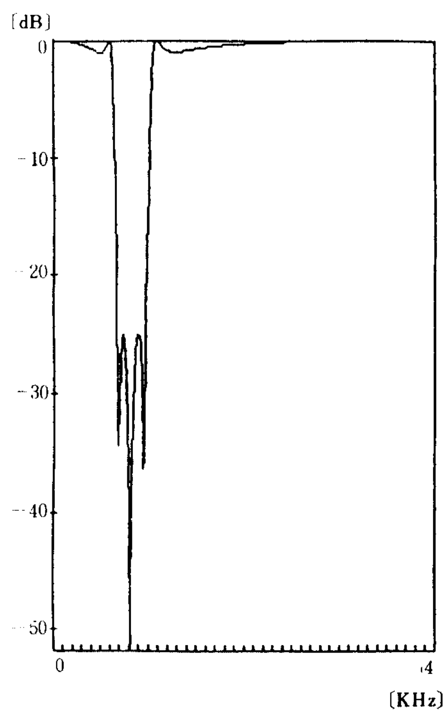


Fig. 3. Frequency Characteristic for BREJ-H (vertical = 10 dB, horizontal = 100 Hz)

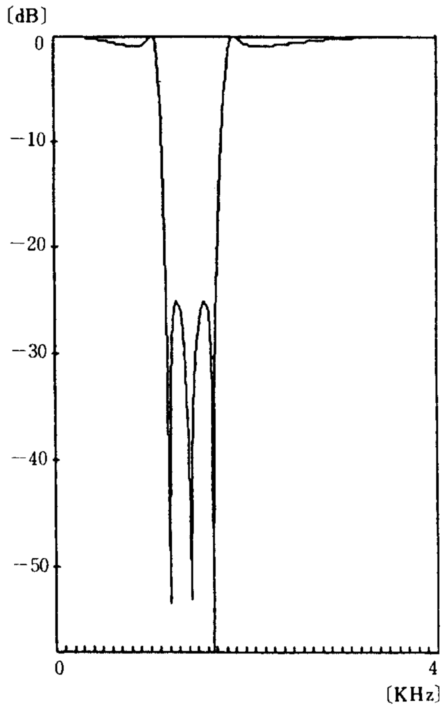


Fig. 4. Frequency Characteristic for BREJ-L (vertical = 10 dB, horizontal = 100 Hz)

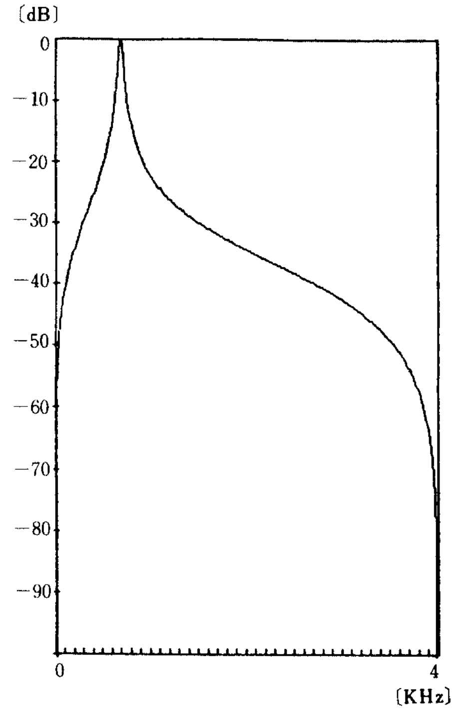


Fig. 5. Frequency Characteristic for 697 Hz RES (vertical = 10 dB, horizontal = 100 Hz)

display for varying input parameters: low group signaling frequency, high group signaling frequency, signal level, dial tone level, gaussian noise level, frequency deviation, twist etc. Fig. 6~20 show the output results for the following input parameters: low group signaling frequency = 697 Hz, high group signaling frequency = 1209 Hz, low group signal level = -10 dBm, high group signal level = -10 dBm, dial tone level = -10dBm, gaussian noise level = -40 dBm (that is, SNR = 33 dB), frequency deviation = 0 %, simulation interval = 50 ms, data interval = 10~40 ms etc. The vertifical scales of the Fig. 6~9 are normalized to the peak value of the input data.

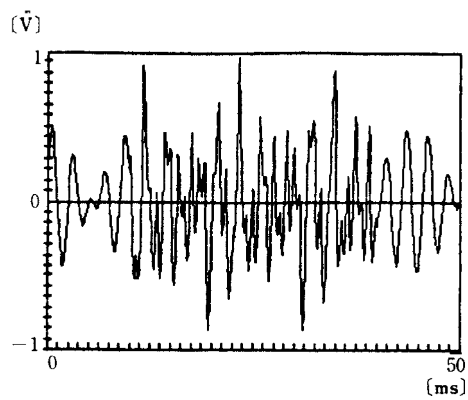


Fig. 6. Input Data (vertical = 0.2 V, horizontal = 1.25 ms)

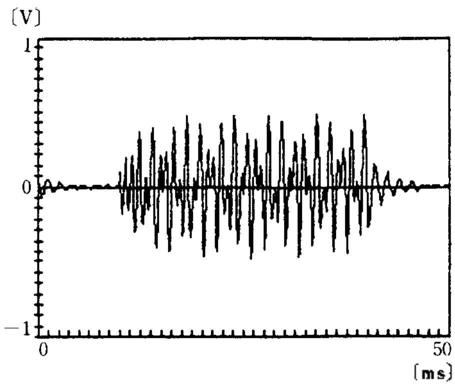


Fig. 7. DTR Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

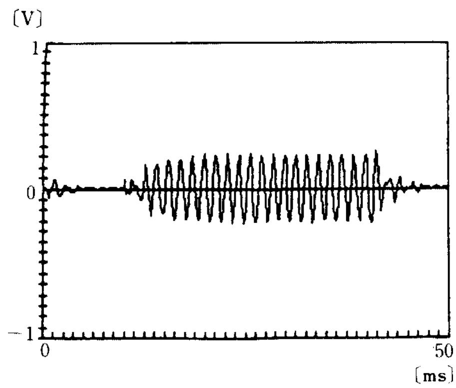


Fig. 8. BREJ-L Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

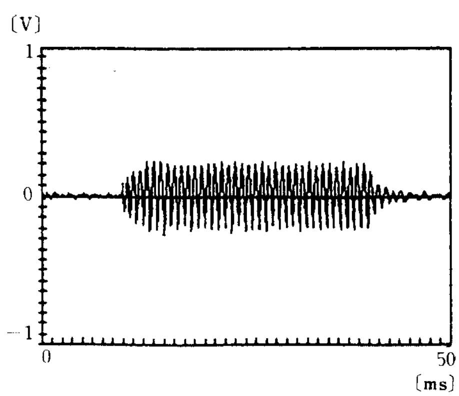


Fig. 9. BREJ-H Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

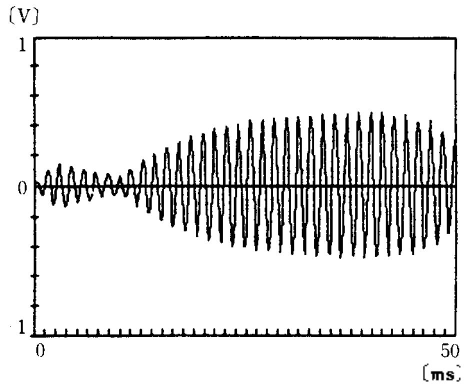


Fig. 10. 697 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

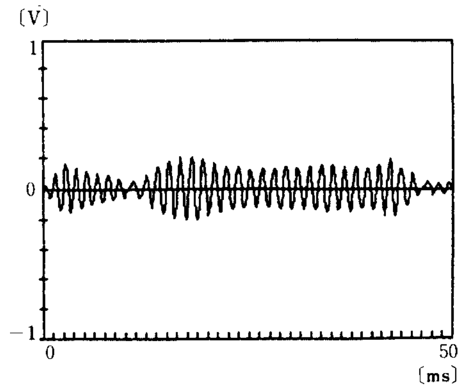


Fig. 11. 770 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

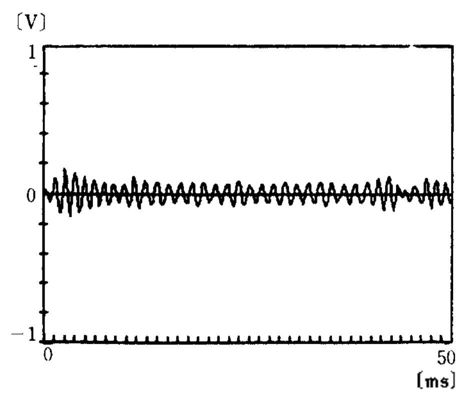


Fig. 12. 852 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

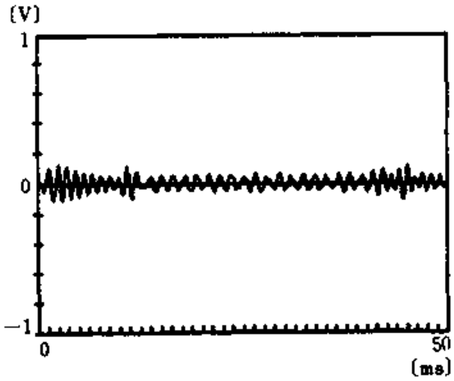


Fig. 13. 941 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

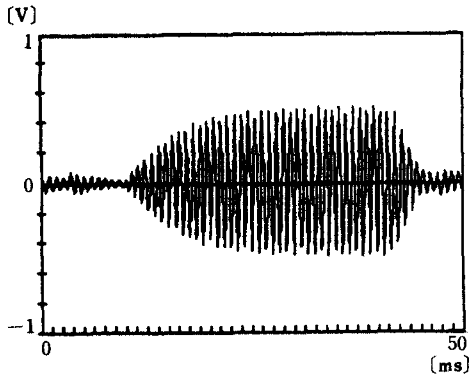


Fig. 14. 1209 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

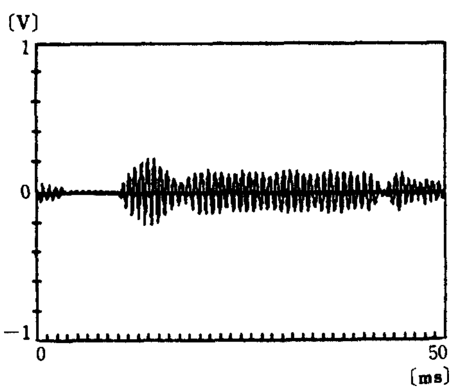


Fig. 15. 1336 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 sm)

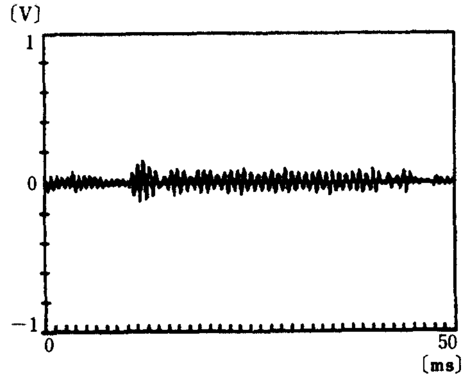


Fig. 16. 1477 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

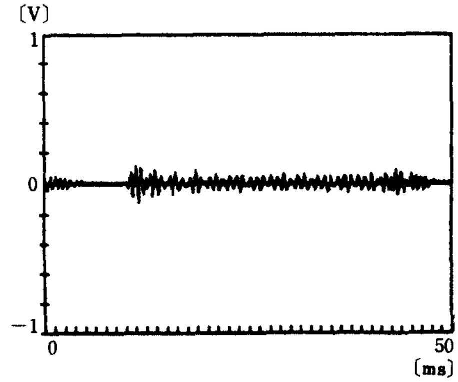


Fig. 17. 1633 Hz RES Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

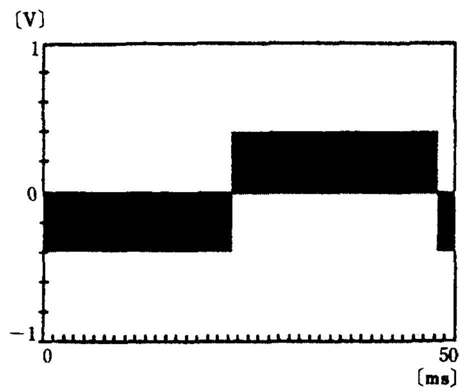


Fig. 18. 697 Hz COM Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

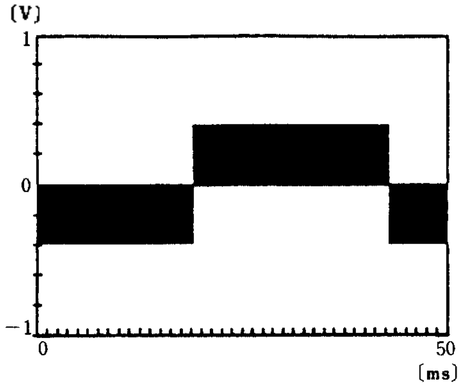


Fig. 19. 1209 Hz COM Output Data (vertical = 0.2 V, horizontal = 1.25 ms)

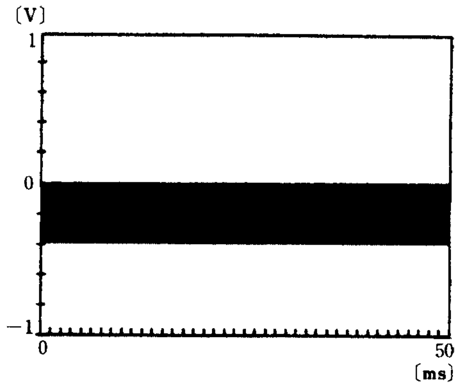


Fig. 20. COM Output Data for Other Frequencies (770, 852, 941, 1336, 1477, 1633 Hz) (vertical = 0.2 V, horizontal = 1.25 ms)

VI. CONCLUSION

In this work we have studied an all-digital DTMF receiver using IIR digital filters. According to simulation results, the receiver meets all the requirements of CCITT recommendations and it is expected that these results can be used without any modifications in implementing the receiver with available DSP LSI.

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