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Efficient Decoding of Digital DTMF and R2 Tone Signalization

Miodrag Popović

Abstract: In this paper the implementation aspects of decoding digital DTMF and R2 signalization tones are analyzed. First, a short review of specifications for DTMF and R2 signalization proposed by ITU-T is made, following by short review of decoding techniques applied in practice. Finally, a new computationally efficient method is proposed and experimentally verified. This method has distinct advantages in multichannel decoding systems and permits decoding in both signaling systems.

Keywords: Signal processing, dual-tone multifrequency signalization, Goertzel's algorithm, QMF filtering.

1 Introduction

Dual-tone multifrequency signalization systems DTMF [1],[2] and R2 [3],[4] are in use more than 50 years since they were introduced in the era of analog telephone system. But, they are also in use today, although tone signals are digital. Early systems for tone detection were based on analog filters. Later, with the advent of digital technology and widespread use of PCM systems, some digital realizations based on the use of digital FIR and IIR filters were proposed [5]-[7]. One of the most used detection systems was based on an algorithm for efficient computation of the Discrete Fourier Transform, known as the Goertzel's algorithm [8],[9]. This algorithm was later improved in

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The author is with the School of Electrical Engineering, University of Belgrade, Bulevar kralja Aleksandra 73, 11120 Belgrade, Serbia and Montenegro (e-mail: pop@el.etf.bg.ac.yu).

some recent papers in order to permit more reliable tone detection [10],[11]. Finally, for the detection of DTMF, a new transform denoted as the Subband Nonuniform Discrete Fourier Transform (SB- NDFT) was introduced, which permits the most efficient computation and reliable signal detection in the presence of noise [12],[13].

The purpose of this paper is twofold. First, it presents a review of existing solutions for DTMF and R2 signaling systems. Second, a new technique will be introduced, which permits the detection of signals in both signalization systems and which is more efficient than existing techniques.

The paper is organized as follows. In Section 2, specifications of DTMF and R2 signaling systems are given, while in Section 3 some existing techniques are explained. In Section 4 the new approach is presented and compared with existing techniques. Finally, in Section 5 some experimental results obtained using Matlab and a system based on digital signal processor TMS320C31 are presented, which show the validity of theoretical predictions.

2 DTMF and R2 Signalizations

In order to make possible interoperability between different networks and between equipment of different manufacturers, the International Telecommunication Union (ITU-T) has been defined a number of standards, known as the Recommendations, which are published after Plenary Assemblies of ITU-T. There are two sets of standards: DTMF standards [1],[2] specifying multifrequency signalization between telephone sets and offices, and R2 standards [3],[4] specifying signalization between telephone offices.

2.1 DTMF signalization

The key technical requirements for the sending and receiving part of DTMF signalization in telephone equipment are defined in ITU-T Recommendations Q.23 [1] and Q.24 [2]. In the sending part, two frequencies are generated, and they are received and detected in the receiving part. Since there are two groups of four frequencies (low and high), 16 valid frequency combinations are obtained, which are presented in Table 1. The following set of frequencies is used for DTMF signalization $\{f_0, f_1, f_2, f_3, f_4, f_5, f_6, f_7\}$ = {697 Hz, 770 Hz, 852 Hz, 941 Hz, 1209 Hz, 1336 Hz, 1477 Hz, 1633 Hz}.

Besides this table of frequencies, the Recommendations [1],[2] specify

Table 1. Symbols and frequencies in DTMF signalization.

f_0	1	2	3	A
f_1	4	5	6	B
f_2	7	8	9	C
f_3	$*$	0	#	D
	f_4	f_5	f_6	f_7

that the decoders are required to detect the frequencies within a tolerance of $\pm 1.5\%$ ± 2 Hz, and not to detect tones that are offset more than $\pm 3.5\%$. The receiver is required to operate within power levels from -2 dBm to -27 dBm and not to operate below -36 dBm. The receiver is also required to detect DTMF signals when two frequencies have different levels (twist) of maximum 6 dB.

The recommendations also specify time constraints for the duration of pulses and pauses that should be recognized or not. The disturbing signals that shouldn't be decoded are also specified. Finally, since DTMF signaling system must work in the presence of speech, the decoder can falsely recognize at most 46 digits during 100 hours of speech at mean power level of -12 dBm.

2.2 R2 signalization

The technical requirements for the sending and receiving part of R2 signalization in telephone equipment are defined in ITU-T Recommendations Q.454 [3] and Q.455 [4]. In the sending part, two frequencies out of possible six are generated, and they are received and detected in the receiving part. There are at most 15 valid frequency combinations, which are presented in Table 2. In some signaling schemes, only 6 or 10 combinations are used. The set of frequencies used for forward direction is $\{f_0, f_1, f_2, f_3, f_4, f_5,\} = \{1380$ Hz, 1500 Hz, 1620 Hz, 1740 Hz, 1860 Hz, 1980 Hz$\}$, while for backward direction the folowing frequencies are used $\{f_0, f_1, f_2, f_3, f_4, f_5,\}=\{1140$ Hz, 1020 Hz, 900 Hz, 780 Hz, 660 Hz, 540 Hz$\}$.

Table 2. Symbols and frequencies in R2 signalization.

f_0	1	2	4	7	11
f_1		3	5	8	12
f_2			6	9	13
f_3				10	14
f_4					15
	f_1	f_2	f_3	f_4	f_5

Besides this table of frequencies, the Recommendations [3],[4] specify that the decoders be required to detect the frequencies within a tolerance of ±10%. The receiver is required to operate within power levels from −5 dBm to −35 dBm. The receiver is also required to detect R2 signals when two frequencies have different levels (twist) of maximum 5 dB for adjacent frequencies and 7 dB for nonadjacent frequencies.

The recommendations also specify time constraints for the recognition of pulses and pauses. The disturbing signals that shouldn't be decoded are also specified.

3 Existing Solutions

3.1 Filtering approach

The filtering approach is conceptually the simplest method for multifrequency detection. The composite signal is usually first filtered by a pair of lowpass/highpass filters and then by a set of bandpass filters as shown in Fig. 1 [5]. The lowpass/highpass group filters divide the composite signal into low and high group in the case of DTMF, and into forward and backward signals in the case of R2. The number of bandpass blocks depends on the number of frequencies in each of the ranges (2×4 blocks for DTMF and 2×5 or 2×6 blocks for R2).

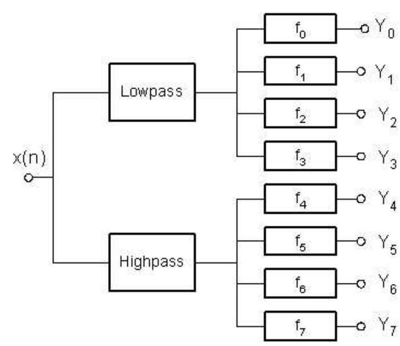

Fig. 1. DTMF detection using digital filters.

The filters used in this approach can be realized as IIR [5], or FIR filters [6],[7]. In the case of IIR realization, the group filters are usually realized as the 4th or 6th order elliptic filters, while bandpass filters can be simpler 2nd or 4th order Butterworth filters. In the case of FIR realization, the group filters are not used, but bandpass filters should be of high order, typically greater than 30.

Due to very high computational complexity, these filters are usually implemented in hardware. Modern tone detection schemes do not use this approach, since it is expensive and it is not suitable for detection of tones in multiple PCM channels.

3.2 DFT approach

The DFT approach is also conceptually very simple. A block of samples should be taken, and its Discrete Fourier Transform should be found for a set of frequencies used in signalization. Since the number of frequencies is small, the use of Fast Fourier Transform algorithms is not suitable. It is known that the most efficient way to find DFT for small number of frequencies is the use of the Goertzels algorithm [8],[9]. The Goertzels algorithm is based on the use of special second-order cell, which uses signal samples as input and gives the DFT coefficients as its output.

The operation of the Goertzel's algorithm is described by two difference equations

$$q_k[n] = x[n] + 2\cos\omega_k q_k[n-1] - q_k[n-2] \quad (1)$$

$$y_k[n] = q_k[n] - q_k[n-1]e^{-j\omega_k} \quad (2)$$

where $x[n]$ are samples of input PCM signals, N is length of the block, and ω_k is the k-th DFT sample.

The block diagram of the Goertzel's algorithm is presented in Fig. 2. The first equation represents the recursive part of the Goertzel's algorithm

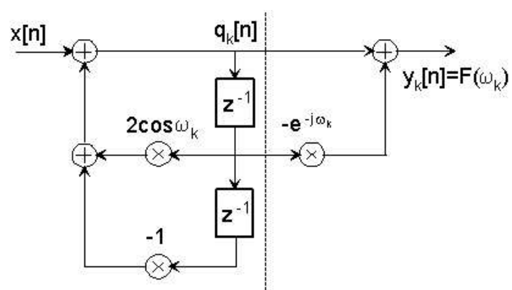

Fig. 2. Goertzel's second-order cell.

(on the left side of dashed line), and it is executed for every input sample, e.g. at sample rate of 8 kHz. The second equation represents the nonrecursive part, and can be executed at N times lower sample rate since

$$X(\omega_k) = y_k[n]\big|_{n=N} = y_k[N]. \tag{3}$$

Therefore, the number of operations for the computation of one DFT sample is $N+2$ real multiplications and $2N+1$ real additions.

The main drawback of this approach is in the fact that frequency bins of a true DFT are evenly spaced

$$\omega_k = 2\pi f_k = 2\pi k f_s/N \tag{4}$$

where N is the length of the block. In general, the DTMF frequencies do not coincide with frequencies at which DFT is computed. These differences can be made small by choosing a large value of N, which is the only free parameter. But, two conflicting criteria should be satisfied. N must be small enough to satisfy time constraints for the detection, and it must be large enough to enable discrimination of DFT samples. Since it is impossible to choose the value of N for which the DFT frequencies exactly coincide with all DTMF or R2 frequencies, it is needed to find the optimal values for N and for set of k, to make these errors smallest. In the best case [8], these errors remain in the range of $\pm 2\%$ of the center frequencies.

This approach was used in many practical implementations of multifrequency tone detectors based on digital signal processors, although it cannot satisfy all requirements in frequency or in time domains, especially in the DTMF detection [8].

3.3 Modified Goertzel's algorithm

The main problem of the Goertzel's algorithm was the difference between the frequencies at which DFT is computed and the frequencies used in DTMF and R2 signalization systems. This drawback was solved in so called the Modified Goertzel's algorithm, first proposed in [10]. Instead of using DFT frequencies (1) and (2), the authors of [10] applied the true DTMF or R2 frequencies to define multiplier coefficients in (1) and (2) and in Fig. 2.

The results of this modification are very good. The center frequencies of the Goertzels cells are coincident with DTMF or R2 frequencies, so that there are no system errors in the computations of tone amplitudes. Further, the value of N can be chosen smaller, so time constraints can be easier satisfied. In fact, the lower limit of N is determined by the acceptable signal to noise ratio of DFT coefficients, and for sampling frequency of 8 KHz, $N = 106$ was found to be acceptable in [10].

The number of operations per one DFT sample is given by the same formulas as in original algorithm, e.g. $N+2$ real multiplications and $2N+1$ real additions, but in this case N is about two times smaller.

3.4 SB-NDFT algorithm

Further improvement in tone detection algorithms was achieved in the Subband Nonuniform Discrete Fourier Transform algorithm [12]. This algorithm is based on the fact that all useful frequencies in DTMF and R2 algorithms are below 2 KHz, what permits subband decomposition and the use of two times lower sample rate (4 KHz). A simple way to get lowpass subband decomposition sequence $g[n]$ is to average successive samples in original sequence

$$g[n] = \frac{1}{2}\{x[2n] + x[2n+1]\}, \quad n = 0, 1, \ldots, \frac{N}{2} - 1 \tag{5}$$

By this addition and decimation by 2, the spectrum of original signal at normalized digital frequency ω_k is

$$|X(e^{j\omega_k})|^2 = \frac{8}{1+\cos\omega_k}|G(e^{j2\omega_k})|^2 = b_k|G(e^{j2\omega_k})|^2 \tag{6}$$

The best way to compute the value of $|G(e^{j2\omega_k})|^2$ is to use the Modified Goertzel's algorithm described in 3.3, but using two times lower sample rate at the input. It should be noted that the Goertzel's cells should be tuned to twice higher normalized frequency ($2\omega_k$) because of decimation of input signal. The block diagram of the SB-NDFT algorithm is shown in Fig. 3.

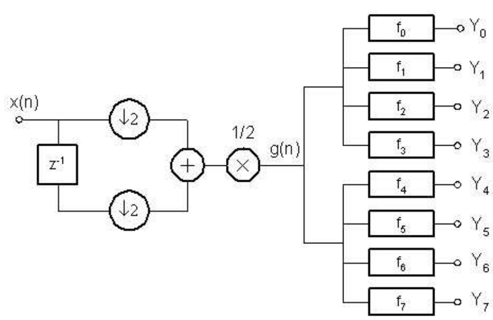

Fig. 3. SB-NDFT detection algorithm.

The quality of signal detection is the same as in the Modified Goertzel's algorithm, but the number of arithmetic operations is almost halved. In

DTMF detection, the number of operations needed to obtain all eight DFT samples is $(4N + 16)$ real multiplications and $(8.5N + 8)$ real additions.

4 New Approach

The use of SB-NDFT permits the most efficient detection of DTMF signals up to date. But, the use the same concept for R2 signal detection is not possible. Namely, although all of R2 tone signals are below 2 KHz, some of possible disturbing frequencies are located above 2 KHz, and they are not sufficiently attenuated by averaging. Therefore, the decoding system for R2 signalization based on SB-NDFT cannot satisfy all requirements proposed in [3],[4].

In order to make possible use of the modified Goertzel's cells at halved sample rate of 4 KHz in the decoding of R2 signalization, some new solution for decimation stage should be found, which should have better attenuation of frequencies higher than 2 KHz and which is still computationally efficient. To satisfy the specifications in frequency domain, the transfer function of this IIR filter must be of seventh order. An efficient way to realize this filter is to use the lowpass part of multirate IIR QMF filter bank [14],[15]. The block diagram of the new algorithm is shown in Fig. 4, while the structure of filtering and decimation stage is presented in Fig. 5.

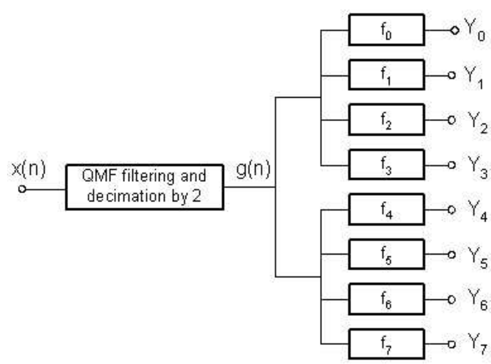

Fig. 4. New detection algorithm.

The lowpass part of the IIR QMF filter bank, shown in Fig. 5, consists of two all-pass branches working at one half of original sampling frequency. Its transfer function can be expressed as

$$H_{LP}(z) = \frac{A_0(z^2) + z^{-1}A(z^2)}{2} \qquad (7)$$

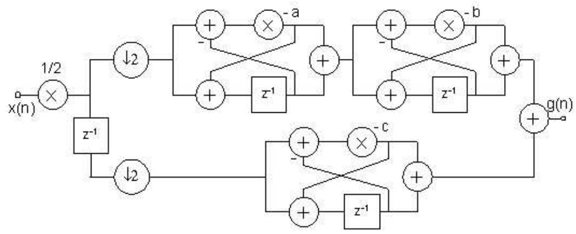

Fig. 5. QMF filtering and decimation.

where $A_0(z^2)$ and $A(z^2)$ are all-pass filter functions realized using half-band IIR filters. It is interesting to note that this filter has only three real multipliers and ten adders, all working at half of sampling frequency. Therefore, this realization requires only 1.5 multiplications and 5 additions per input sample.

The total number of arithmetic operations is moderately higher than in the SB-NDFT approach. The complete realization of eight frequencies DTMF decoder requires $(5.5N + 16)$ real multiplications and $(13N + 8)$ real additions per block. This should be compared with $(4N + 16)$ real multiplications and $(8.5N + 8)$ real additions in the SB-NDFT approach, or with $(8N + 16)$ real multiplications and $(16N + 8)$ real additions in the Modified Goertzel's approach. But, the big advantage of new approach is that it can be used for both DTMF and R2 signalization, while the SB-NDFT and the Modified Goertzel's approach cannot be used for R2 decoding.

5 Experimental results

The proposed algorithm for DTMF and R2 signal detection was simulated in Matlab programming environment, and a series of tests was performed.

Preliminary tests for DTMF decoding used artificially created tone signals, which simulate different conditions proposed by specifications [1],[2]. Final tests for DTMF decoding have been performed using a digital version of the special purpose Mitel test cassette CM7291 [17], which contains recorded real tone DTMF signals as well as real disturbing signals. All performed tests (recognition bandwidth, center frequency offset, twist, dynamic range, time constraints, acceptable signal to noise ratio, and influence of speech) passed successfully. It is interesting to note that our new algorithm detected only 10 symbols in 30 minutes of condensed speech, although up to 30 false detected symbols are acceptable.

All tests for R2 decoding used artificially created tone signals, which simulate different condition proposed by specifications [3],[4]. All performed tests passed successfully.

Finally, the algorithm was realized on a DSP platform based on TMS320-C31 floating-point digital signal processor. It could successfully decode signals from 32 PCM channels in real time, using only on-chip memory. This board is in production and several hundreds of boards are built-in into domestic DKTS telephone exchanges.

6 Conclusion

This paper is devoted to problems and solutions of detection of dual-tone frequency signalization used in telephone exchanges. A review and discussion of modern approaches was performed. Finally, a new realization is described, which permits detection of both DTMF and R2 signaling tones and which is computationally efficient. The results of simulations and real measurements are presented.

References

[1] International Telecommunication Union, ITU-T Recommendation Q.23, "Technical features of push-button telephone sets." Fascicle VI.1, Blue Book, 1993.

[2] International Telecommunication Union, ITU-T Recommendation Q.24, "Multifrequency push-button signal detection." Fascicle VI.1, Blue Book, 1993.

[3] International Telecommunication Union, ITU-T Recommendation Q.454, "The sending part of the multifrequency signaling equipment." Fascicle VI.4, Blue Book, 1993.

[4] International Telecommunication Union, ITU-T Recommendation Q.454, "The receiving part of the multifrequency equipment." Fascicle VI.4, Blue Book, 1993.

[5] S. Kikuchi, H. Kitano, and J. Matsuda, "A study on digital pushbutton signaling receiver," *Review of Electr. Commun. Lab.*, vol. 27, no. 7-8, pp. 630–641, Aug. 1979.

[6] F. Braun, "Nonrecursive digital filters for detecting multifrequency coded signals," *IEEE Trans. Acoust. Speech, Signal Process.*, vol. 23, no. 3, pp. 250–256, June 1975.

[7] G. Thyssens and L. Verbist, "Digital multifrequency receivers and senders," *Electr. Commun.*, vol. 54, no. 4, pp. 319–325, 1979.

[8] I. Koval and G. Gara, "Digital MF receiver using discrete Fourier transform," *IEEE Trans. Commun.*, vol. 21, no. 12, pp. 1331–1335, Dec. 1973.

[9] P. Mock, "Add DTMF generation and decoding to DSP μP designs," *EDN*, vol. 30, pp. 205–220, March 1985.

[10] S. Gay, J. Hartung, and G. Smith, "Algorithms for multichannel DTMF detection for the WE® DSP32 family," in *Proc. IEEE Int. Conf. Acoust. Speech, Signal Processing, Vol. 2*, (Glasgow, UK), pp. 1134–1137, May 1989.

[11] A. Dabrowski, "Modified Goertzel algorithm for detection of dtmf signals," in *Proc. Int. Conf. on Signal and Systems*, (Algiers, Algeria), pp. IV.1–IV.3, Sept. 1994.

[12] S. Bagchi and S. Mitra, "The nonuniform discrete Fourier transform and its applications in filter design: Part 1–1-D," *IEEE Trans. Circuits Syst. II: Analog and Digital Signal Processing*, vol. 43, pp. 422–433, June 1996.

[13] S. Bagchi and S. Mitra, "Efficient robust DTMF decoding using the subband NDFT," *Signal Processing*, vol. 56, pp. 255–267, 1997.

[14] Lj. Milić, M. D. Lutovac, and D. Ilišević, "Characteristics of two-channel IIR QMF banks," in *Proc. TELFOR-2000*, (Beograd, Yugoslavia), pp. 333–336, Nov. 2000.

[15] Lj. Milić and M. Lutovac, "High speed IIR filters for qmf banks," in *Proc. 5th Int. Conf. TELSIKS 2001*, (Niš, Yugoslavia), pp. 171–174, Sept. 2001.

[16] M. D. Lutovac and L. Milić, "Half-band IIR filter design using MATLAB," in *Proc. TELFOR-2000*, (Beograd, Yugoslavia), pp. 329–332, Nov. 2000.

[17] Mitel Semiconductor, "Tone receiver test cassette CM7291." Mitel Technical data manual, San Diego.