Considerations for DTMF Generation with Micro-Controller System

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Abstract - **The micro-controller unit (MCU) is presents in many system connected to the telephone network . Sometimes MCU has free resources which could be used for additional functions. It is possible to use them for standard or user-defined single, Dual Tome Multi Frequency (DTMF) or multi-tone (MT) generation and recognition.**

Keyword - **DTMF generation, digital processor, micro-controller**

I. Introduction

The micro-controller (MCU – Micro Controller Unit) is present in many systems connected to the telephone network and some of its resources are free to be used for additional functions. Consequently it is possible to use them for analog signals (AS) generation and recognition and in particular for standard or user-defined single or Dual Tome Multi Frequency (DTMF) generation or for melody and other audio signal generation. Most of the general purpose MCUs such as Motorola 68HC11F1 [1] and Microchip PIC 18F876 [2] could be used successfully to generate audio signals. A lot of specialized integrated circuits designed to generated and recognize standard DTMF tones exists [e.g. 3, 4] and could be used but sometimes they are additional burden to the designer and the manufacturer of the system. In the privately owned communication systems an original one or multi-tone signaling system could be used.

The purpose of this paper is to apply and test the methods in the signal sampling and reconstruction described in [5, 6] in the field of DTMF generation and recognition by digital processor system and in particular by MCU-driven system.

DTMF generators are already implemented in some MCUs. For example in MC68HC05F32/F8 there are implemented DTMF/Melody Generators with signal sampling factor (SSF) N=Fd/Fs=28 and with 6-bit internal D/A converter [9] .

The value for N and n in these and many more cases are in relative good correspondence with the equations discussed in [5, 6] and given below:

$$n=\lg((N))+2=\lg(28)+2=6.81 \text{ bits} \qquad (1)$$

and with the Eq. 2

$$N=\lg((1/E_{MAX}))=\log((1/0.00629))=7.32 \text{ BITS} \qquad (2)$$

where logarithm in base 2 and Emax=0.629% for N=28.

In some cases using the MCU with build-in DTMF generator decoder is not possible and the only solution is to implement these functions on the general purpose processor already available in the project.

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II. DTMF tones

The DTMF and call progress frequency list is given in Table I [3,4]. The solutions for DTMF generation discussed here are applicable for any the audio-tone and melody generation system and are interesting for different purposes (medical, security, voice synthesis, testing electronics equipment and communication channel etc).

TABLE I.
DTMF AND CALL PROGRESS TONES.

Description	Tone 1, [Hz]	Tone 2, [Hz]
Dial Tone	440	350
Busy	620	480
Ring-back	480	440
1	1209	697
2	1336	697
3	1477	697
4	1209	770
5	1366	770
6	1477	770
7	1209	852
8	1366	852
9	1477	852
0	1366	941
*	1209	941
#	1477	941
A	1633	697
B	1633	770
C	1633	852
D	1633	941

Although the sampling frequency Fd of approximately 8 KHz is used in some of the telephone connected equipment (Table II) this not a strict rule and the designer could take any sampling frequency Fd in order to solve the problem. But if the sampling frequency Fd is kept constant the amplitude error Emax is different for each frequency Fs of DTMF signals (Table II).

It is seen that DTMF tones are falling into two main groups: 1/ low group: 697, 770, 752 and 941 Hz and 2/ high group: 1209, 1336, 1477 and 1633 Hz. The “dead band” between the groups is 268Hz with central frequency of 1075Hz.

Note: 1/ With F=8kHz N=Fd/Fs>=4 for all frequencies and Eamax<=-3dB. 2/ Emax is a maximal amplitude error due to he possible sampling the SS/CS not in its maximum.

TABLE II
SAMPLING DTMF SIGNALS WITH FD=8KHZ

Low band (ΔF=244Hz)				High band (ΔF=424Hz)		
Fs, Hz	N	Emax		Fs, Hz	N	Emax
697	11.48	3.72%		1209	6.62	11.1%
770	10.39	4.53%		1336	5.86	11.4%
752	10.64	5.55%		1477	5.42	16.4%
941	8.50	6.75%		1633	4.9	19.9%

The following general rule (GR) for adding SS/CS was formulated: *1/ When k SS/CS signals with the same amplitude A are added the amplitude of the resulting signal could be up to k*A and the peak-to-peak amplitude of the same signal could be up to 2*k*A. or 2/. When k SS/CS are added the amplitude of the resulted signal cannot exceed the sum of the amplitudes of the signals in the sum.*

The major consequence of this GR is that the resolution and the accuracy of each signal component in the sum is reduced in comparison to the resolution and accuracy of the output signal. For example if an 8-bit DAC is used and only one SS/CS is generated it could have full 8-bits of accuracy and resolution (256 steps). If two tone are generated simultaneously each of them could have up to 7 bits (128 levels), if four tones are generated simultaneously they could have 6 bits (64 steps), etc and the Eq. 3 is applicable

$$Bs = \lg (Lv/k), [bit] \qquad (3)$$

where Bs are bits per signal component, Lv a levels in the staircase function of the converter (Lv = 2exp(n) or Lv = 256 for a n=8 bit converter) and k is the number of the tones in the sum (Table III). Also, the signal to noise ration (SNR) for each individual tone is reduced when the number of SS/CS k is increased and the number of bits n is kept constant.

In some case in order to keep the reconstructed SS/CS fully symmetrical (to zero the direct current (DC) offset) *the number of the signal levels is reduced by one* (for example 255 levels or steps are used instead of 256 for an 8-bit DAC with middle level at 127th step or 127 (+127, -127) steps and not at 127.5 (+127.5, -127.5) or at 128 (+127,-128) steps. The table below (SIN16_6) is an example with codes for SS generation with SSF N=16, n=6 bit and 64 levels from 64. The middle level is Lmiddle=31.5, the minimal level is Lmin= 0, the maximal level is Lmax=63, the resolution in phase is LSBph=360/16=22.5 deg., the asymmetry (the difference between the positive and the negative amplitude) is 1*LSB and the angle of the first sample is φ0=0deg.

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SIN16_6   FCB  31, 44, 54, 61, 63, 61, 54, 44
          FCB  31, 19, 09, 02, 00, 02, 09, 19
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Some amplitude optimization (or exemption of the GR given above) could de done during the DTMF generation. This is possible because the frequencies of the DTMF tones are known and the phase relations between the two tones are controllable and the duration of the tones are also known. Consequently the maximum peak to peak value of the sum of two fully defined SS/CS could be less that the sum of the two peak-to-peak value and this is predictable.

Another consideration is the minimum amplitude value of the sum of two or more SS/CS. It is possible that the sum of two signals to become too small (comparable to the value of one least significant bit (LSB)). It is suggested that the smallest AS amplitude to be at least 10*LSB (or to have at least 3 bits of value) in order to be reconstructed with good THD, IMD and SNR and relative weight of one LSB to be less than 10%. The full scale (FS) range of the converter is not always possible to use in order to avoid the under-voltage and over-voltage which are leading to he irreversible lost of information.

TABLE III
NUMBER OF SIGNALS K AND THEIRS PEAK-TO PEAK AMPLITUDE WHICH COULD BE GENERATED WITH AN 8-BIT DAC.

k, Nb of signals	Parameter per signal			
	n, bits	Lv, Levels per signal	SNR, dB	Nmax, (Fd/Fs)
1	8	256	49.92	64
2 **	7**	128**	43.9**	32**
4	6	64	37.88	16
8	5	32	31.86	8
16	4	16	25.84	4
32	3	8	19.82	2
64	2	4	13.8	2*
128	1	2	7.78	2*

Notes: *- conditional value; ** - for DTMF

Two approaches for DTMF and multiple tone generation are evaluated:

1. Using one sampling frequency Fd=constant for generating all tones (for example with N=Fd/Fs>=4) and different SSF N=Fd/Fs for each tone in the sum.
2. Using one SSF N=Fd/Fs for all frequencies, and different sampling frequency for each SS/CS in the DTMF (sampling frequency Fd=variable).

The second approach was retained because only one sine table is needed for all tones and only one evaluation of the spectrum of the produced SS/CS approximation is needed. The disadvantage is that multiple sampling frequencies Fd should be generated. Usually this is not a difficulty for most of the MCUs where several timers are available or could be added at low cost and a powerful interrupt system exists.

In order to produce a DTMF or any dual-tone AS two main methods are used:

1. Generating two individual AS and summing them.
2. Digitally summing the samples of the two AS and using one DAC to produce the analog function.

The first approach is considered better because each output signal could be controlled individually, but the second approach is simpler and cheaper.

The SNR for each individual SS/CS in the sum could be theoretically calculated according to the Eq. 4

$$SNR=6.02*n+1.76, [dB] \qquad (4)$$

Where n is the number of bits used to represent the peak to peak amplitude of the tone.

III. CHOOSING THE SAMPLING FREQUENCY FD AND NUMBER OF BITS PER TONE N

According to the Nyquist criteria and the sampling theorem of Whittikar-Shannon-Kotelnikov the sampling frequency Fd should be at least two times higher than the synthesized tone Fs (or Fd>=2*Fs). We will abandon this criteria and the theorem because they have the followings weak points and are unpractical:

1. They do not cover the cases when the AS could be constructed with SSF N=Fd/Fs<2.
2. They do not give the possibility to calculate the minimum number of bits n into the digital word used to represent the analog signal.
3. They do not evaluate the errors and the quality of the produced signal (amplitude errors, DC error, THD, IMD, SNR, etc.)

Since the needs of the nowadays DTMS standards are largely satisfied with SSF N>8 and n>= 6 bit we will use the approach defined in [5, 6] and based on the Eq 5 and Eq. 6 for the maximal amplitude error Emax, SSF N=Fd/Fs and the recommended minimum of the converters bits n(dac):

$$Emax = 1-\sin(90-180/N)), [-] \quad (5)$$

For N= 2 to 48 the approximate Eq. 6 is applicable.

$$n(dac)>=lg(N)+D, \text{ bits} \quad (6)$$

For any value of N>=2 Eq. 7 is applicable

$$n(dac)>=lg(1/Emax)+D, \text{ bits} \quad (7)$$

where lg is in base 2, and D is a constant from 0 to 4, depending on the application. If we are using the formula above we could calculate the maximum recommended SSF Nmax when the number of bits n are known and vice versa. Using higher SSF than the calculated Nmax for the selected number of bits n is possible but not always useful due to the rounding errors. Table IV is giving sampling frequencies for DTMF generation with N=8 and 16. Musical notes also could be generated (Table V).

TABLE IV
SAMPLING FREQUENCIES FOR DTMF FREQUENCIES WITH N=FD/FS=8 AND N=FD/FS=16

Tone, Hz	Fd (N=8), Hz	Fd(N=16) , Hz
697	5576	11152
770	6160	12320
752	6816	13632
941	7528	15056
1209	9672	19344
1336	10688	21376
1477	11816	23632
1633	13064	26128
n, bits	>=5	>=6

TABLE V
SAMPLING FREQUENCIES FOR THE MUSICAL NOTES FROM OCTAVE 4 WITH N=FD/FS=8 AND N=FD/FS=16

Note	Tone, Hz	Fd (N=8), Hz	Fd(N=16) , Hz
C	261.63	2093.04	4182.08
C#	277.18	22.17.44	4434.88
D	293.66	2349.28	4698.56
D#	311.13	2489.04	4978.0
E	329.63	2637.04	5274.08
F	349.23	2793.84	5587.68
F#	369.99	2959.92	5919.84
G	392.00	3136	6272
G#	415.30	3322.4	6644.8
A	440.00	3520	7040
A#	466.16	3729.28	7458.56
B	493.88	3951.04	7902.08
n, bits		>=5	>=6

IV. MAIN PROBLEMS TO BE SOLVED

The following questions should be solved in order to generate successfully AS with digital processor system:

1. Choosing the method for signal reconstruction. Usually the choice is between the pulse width modulation (PWM) and direct signal synthesis with DAC.
2. The followings methods for samples calculation are possible 1/ to calculate the samples of the complex AS and using one DAC or 2/ simultaneously to generate with two DACs two separate AS and summing them in an analog system or 3/ to generate two separate samples from two different AS and summing them digitally in one DAC.
3. Choosing the size of the tables. If the same SSF (e.g. N=16) for generating all DTMF signals is used, only one sine table is needed with N entries
4. Choosing the length of the digital word n which if necessarily but usually multiple of 8 bits. It depends on the SSF, the errors and the signal to noise ratio (SNR).
5. Choosing the method of the amplitude correction, in any. For example in some systems it is necessarily to amplify additionally the higher frequency signal. This could be done either by multiplying the amplitude of the higher frequency samples or by reducing the amplitude of the low frequency samples or both of them in the same time. The multiplication and the division could be done with shifting, adding and subtracting. The correction of the amplitude depends on the parameters of the communication channel and the SSF N. If the same SSF is used for all generated SS/CS the same maximal amplitude error Emax is maintained and the amplitude correction could be avoided. In all cases the signals should be kept in the allowed range.
6. Choosing the type of the analog filters after the DAC. In most of the cases it is obligatory to use first or maximum a second order passive low pas filter in order to keep the production cost low. In order to keep the filter as simple as possible it is necessarily to use the maximum SSF N

(preferably N>8 for a SS) and the maximum number of bits according to the Eq. 5 or Eq. 6 and preferably n>=5 bits. In most of the cases N>=16 and n>=6 bits are giving excellent results for DTMF.

7. Switching On/Switching Off problem during AS generation should be resolved. In order to minimize the THD the synthesis of each tone should be started from amplitude zero and with minimal slew rare (SR) and should be ended in zero (when is possible) and whole number of periods of each signal should be generated. Normally the switching ON process is generating more harmonics than the switching OFF.
8. An alternative approach with digital signal processor (DSP) was evaluated and was considered as inappropriate either because the addition of a DSP is not possible in many cases or DSP cannot be used at the moment to solve the tasks implemented in a MCU.

V. CONCLUSIONS

The methods described in [5, 6] were applied for a MCU-based system for single or multiple tone (sine wave) generation and results where obtained. Solutions for MCUs Motorola 68HC11F1 and Microchip PIC16F876 were developed but the paper is applicable of any system based on a digital processor.

It is considered that *any real signal extend in an finite frequency range due to its finite slew rate (SR), finite peak to peak amplitude (App) and finite power (Ps). Real signals are generated and transmitted in the components with finite frequency characteristics.* It is wrong to imagine signals with infinite SR and "ideal angles" as the "ideal square wave signal" and to apply to them the Furrier transform which is a good solution only for signals with finite SR. If this application is done an impossible to correct error will be generated due to the different nature of the "ideal" signal and the Fourier transform.

In most of the cases increasing the SSF N and the corresponding increasing of the number of bits n in the digital word is the perfect solution in order to enhance the quality of the signal generation and processing. For example choosing a higher value for the SSF N will lead to simpler filter after the reconstructing DAC and to less THD and IMD.

The AS (even a simple one as a SS/CS) is containing an infinite quality of information which could be only partially represent in a digital form. In order to represent the AS as accurate as possible in the digital form the highest possible values for N and n should be chosen.

The THD of the reconstructed SS/CS and the IMD of the sum of two or more SS/CS are directly related to the chosen SSF N and number of the bits n. Two approached were tested:

1/ fixing N and changing n; 2/ fixing n and changing N.

Is was found that there is a practical optimal set of values for N and n. Consequently for a fixed value of N in most of the cases there is no practical use to enlarge n more than the values given with Eq. 5 or Eq. 6. The reverse is also true.

In general when the AS is sampled two parameters are changed: 1/ the time is changed from analog to discrete
2/ the amplitude is changed from analog to discreet.
In order to analyze the situation the approach "One change at a time" was used. The amplitude should be changed to discrete and time is kept analogous and vise versa.

One more problem with the accuracy of SS/CS signal reconstruction is that the signals parameters are changing with changes of the signal amplitude. For example the amplitude value of SS at +90 degree is coded with maximal accuracy and the value of +270 degree with minimal relative accuracy, which is a source of additional signal distortion and could be investigated.

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