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DTMF using software

The M16C forces its way into the field of DSP applications

Compared with digital terminals, analogue telephones are an attractive alternative. In addition to the price advantage, they offer a number of integrated convenience features. For this purpose DTMF signals, among others, are employed. Usually they are generated and detected by special components or DSPs. However, software can also be used on a powerful microcontroller. With reference to a complete software application to detect DTMF tones, this article reveals the suitability of the M16C family from Mitsubishi for digital signal processing.

DTMF stands for "Dual Tone Multi Frequency" (multi-tone dialling procedure). For this dialling procedure the telephone keypad is regarded as a 4X4 matrix and each column and each row is allocated to a particular frequency **[1]**. The 4 frequencies in the rows form the lower frequency group and the four in the columns form the upper group. Therefore, each key is represented by a unique combination of two frequencies **(Figure 1)**.

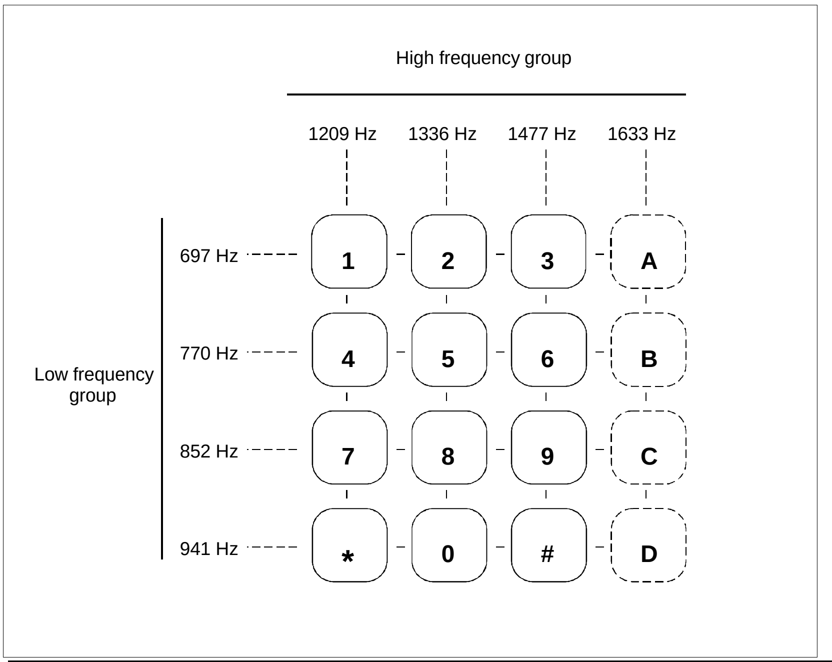

Figure 1: The allocation of frequencies in the framework of the 4x4 matrix on the telephone keypad

Today DTMF signals are not only used for connection setup in analogue telecommunication, but also for remote control of telephone answering machines, for communication with automatic dialling systems, for selecting services from private branch exchanges and in some countries (e.g. Denmark) also for caller identification (Caller ID).
Whereas the generation of these frequencies is relatively straight forward using software, detection represents a greater challenge. Ultimately a decision has to be made in real time whether the signals received are valid or whether they are merely an audio signal with similar spectral parts, for example. The DTMF tones generated do not generally reach the recipient in its original form. The level of the signal is attenuated by the transmission impedance (frequency dependant). The

proportions of noise add up. The tone may also be interrupted temporarily by line disturbance. All of these parameters and their permissible thresholds are laid down in national and international standards **[2]**.

Special hardware is inflexible, expensive and requires additional PCB space

The software applications **[3, 4]** available via the Internet are almost exclusively DSP implementations and only marginally touch on conformity with international (ITU-T) or national standards. This is the crux of the actual problem. A report compiled by Berkeley University **[5]**, in which various DTMF algorithms were investigated and compared, makes clear that even on DSPs this task is not child’s play. As a rule, hardware in the form of special telecommunications chips is still being used. The disadvantage: too inflexible, additional board space required, additional costs.

M16C with special instructions

In this article, the M16C microcontroller family from Mitsubishi Electric has been selected as the hardware platform. The M16C has special instructions at its disposal (e.g. RMPA - repeat multiply and add) suitable for this application as well as extensive peripherals (e.g. 16bit*16bit multiplier and DMA channels). Various derivative versions with different ROM/RAM configurations on the chip are available to the developer. The flagship is currently the M16C/62 with a minimum instruction execution time of 62.5ns and a processor power of 8 VAX-MIPS at 16MHz-clock frequency. This CPU version is available with 256 Kbyte Flash and 20 Kbyte RAM. **Figure 2** shows the internal configuration of the microcontroller. Its bigger brother is close at its heels: The M16C/80 is advancing strongly in the direction of 32-bit processors and handles up to 15 VAX-MIPS at 20MHz.

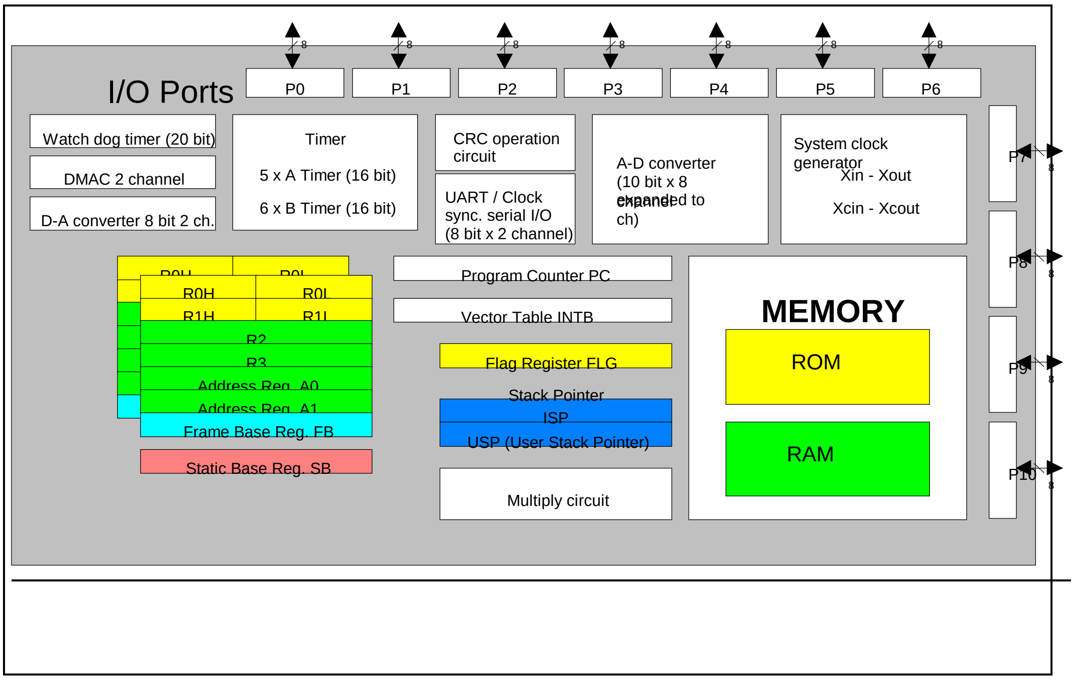

Figure 2: The M16C/62 handles 8 VAX-MIPS at a clock frequency of 16MHz. The minimum instruction execution time is 62.5 ns.

DTMF algorithms in software

The algorithm for detecting DTMF signals is configured to handle the following subtasks: Detection of signal start and end, processing the values of the A-D converter in digital filters as well as evaluation of the filter outputs and interval times in a decision logic.

For pure filtering, algorithms, which exploit the fact that the sought frequencies are already known, are particularly suitable. These include, in particular, the well-known Görtzel filter **[6]** and the wave digital filter (WDF) **[7, 8]**, which is based on the resonance principle. The computation requirements are roughly comparable, the WDF requiring fewer samples, but more operations. However, the WDF dropped down significantly in the Talk-Off test (see below) and clearly exceeded the permissible thresholds.
However, the Görtzel filter, based on a discrete Fourier transformation, only supplies the exact energy value for one spectral line. The maximum output energy is output if the incoming signal corresponds exactly with the frequency of the filter. If the incoming frequency deviates from the expected value, the energy falls. In order to establish the frequency deviation, the input amplitude of the signal frequency must be set in relation to the output energy of the filter. However, the input amplitude now comprises the sum of both DTMF frequencies. The level of both received signals can vary greatly. This is why both frequency groups (columns and rows) must also be separated from each other by means of a high-pass filter or a low-pass filter. A FIR filter 15^{th} order was used for this purpose. It is important to consider that the FIR filter also attenuates or amplifies the signal within a frequency group dependent on frequency. Although the relation between the input amplitude of a frequency and the output energy has now been determined, it must be dealt with separately for each frequency. This problem could be solved to the greatest possible extent using a window function. However, this would require significantly more computation time and memory.

The last A-D values are buffer stored for detection of the signal start and end. At the same time, this array serves as delay for the FIR filter. If these stored values do not exceed a dynamically defined noise threshold, this is recognised as signal interruption. A timer assists in differentiating between tone interruption and tone end.

A signal start is detected as soon as a number of subsequent samples lie outside the noise. The decision logic establishes which Görtzel filter of the lower frequency groups and which of the upper frequency groups produces the maximum output energy. A series of limiting values intercept "impossible" combinations. Therefore, a minimum energy must be reached, for example, so that the signal is not discarded. The output energy to be expected can be calculated from the output amplitude of the FIR filter using a quadratic equation. If the value falls significantly below the desired energy, a frequency deviation is assumed. The advantage of this procedure is that the parameters can be adapted to specific requirements and /or standards.

During implementation, a sample frequency of 8kHz was selected. Tasks, such as FIR filtering and also the recursive part of the Görtzel filter, are carried out within the A-D interrupt routine. This enables you to achieve online processing in real time. Therefore, a maximum of 125 µs minus the interrupt latency are available. The decision logic is activated in the main loop after 102 samples. A signal must then be accepted in a raster of 12.75 ms on at least three consecutive occasions in order to represent a valid key.

There are few requirements of the hardware. The level of the signal must be shifted to Vcc/2. In addition, a standard low-pass filter is connected in order to limit the bandwidth to the 3.4 kHz typical of telecommunications. The OpAmp used and the MCU are supplied with the same voltage (5V), which severely restricts the amplitude range available. The exact trigger of the A-D converter is realised via a timer output, which is directly linked to the A-D trigger input. For test purposes, an LED array and a serial interface were also used on the hardware. Thus, key detection can be traced and logged conveniently via a terminal emulation. The whole configuration of the hardware and software is shown in **Figure 3 [9]**.

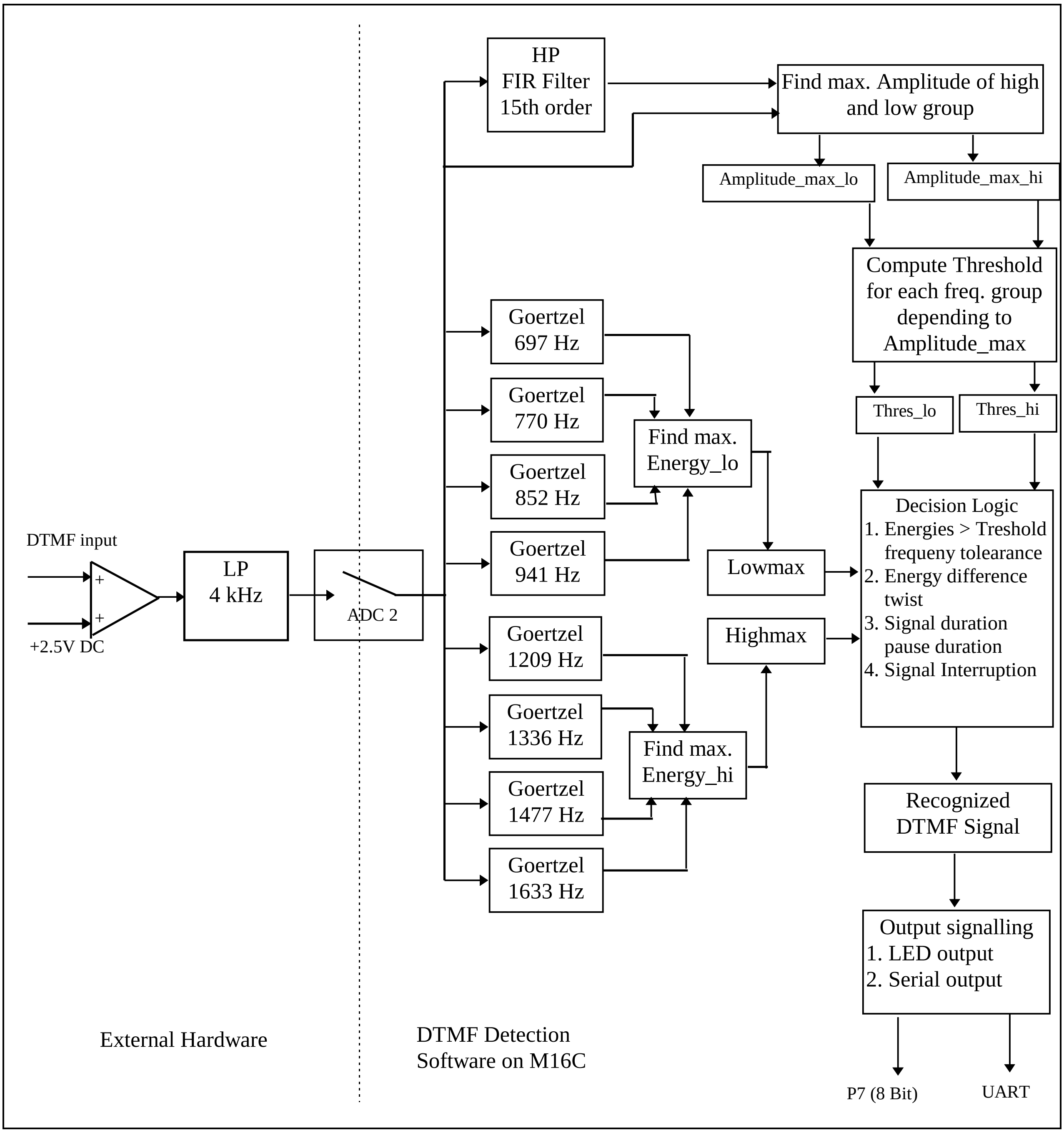

Figure 3: Software concept and hardware interface connection

The Mitel audio test tape was selected for the test. To improve handling, the MC tape was digitised and processed with the aid of CoolEdit® software from Syntrillium Software® and played using a commercial soundcard. In this way the tests could be configured and additional ones generated with ease. **Table 1** shows the results.

	Description	Test results
Test #2 (*) Basis test	All samples are each generated ten times with a signal-pause ratio of 50 ms to 50 ms.	All 120 samples are detected
Test #3 Frequency tolerance test	The frequency of a frequency group is varied from -4% to +4% in 0.1% steps, while the frequency of the other groups remains constant.	Up to at least 1.8% frequency deviation, the sample is detected reliably. Depending on the sample, the signal is discarded as of a frequency deviation of 2.2% to 4%.
Test #4 Amplitude ratio	The amplitude ratio between the two frequency groups is varied between 0db and 10dB.	>8dB
Test #5 Level	The signal level is varied by 30dB in steps of 1dB.	0dB - 28dB
Test #6 Minimum signal length (Guard Time)	A range of tone pulses is generated, which are shortened continually.	38 ms
Test #7 Signal noise spacing	White noise is added gradually to the actual signal at -24dB, -18dB and -12´dB. 1000 tone pulses are sent in every stage.	All 3000 pulses are detected
Test #8 Talk Off Test	Speech and other signals typical to telephones are sent for half an hour. The sequences were selected from the standpoint of being able to detect mismatched detections easily.	Only 4 mismatched detections detected

(*) Test #1 does not exist

The test results show that the specifications of several standards can be fulfilled. The parameters can be varied within certain limits. However, the maximum possible twist (The "twist" describes the difference in level between the two components of the DTMF signal), for example, is also reduced when the permissible frequency tolerance is limited. If the level difference increases, more mismatched detections are detected in the Talk-Off test etc. The 16-bit fixed-point arithmetic also sets natural limits. In particular, maximum frequency tolerance, which is not even required in some standards or is significantly higher, requires considerably higher expenditure. Without this limitation, the FIR filter, for example, would not necessarily be required and the decision logic would be simplified. The hardware interface connection also offers scope for optimisation. The OpAmp limits the signal deviation to Vss=2.6V and thus only half of the 10bit A-D converter range is used. Table 2 gives an overview of the resources of the M16C for complete conversion.

Processor load	M16C/80: 23% at f(xin)=20MHz, Vcc = 5V M16C/62: 42% at f(xin)=16MHz, Vcc=5V M16C/61: 67% at f(xin)=10MHz, Vcc=5V	During an applied signal
ROM	approx. 4.9kB	excludinginitialisation and test routines
RAM	approx. 0.9kB	
Peripheries	3 Timer, 1 DMA channel, 1 A-D channel	Reduction to 1 HW timer, 0 DMA, 1 A-D channel possible with more software

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