AN616

Digital Signal Processing with the PIC16C74

Author: *Darius Mostowfi*
Design Consultant

INTRODUCTION

This application note describes the basic issues that need to be addressed in order to implement digital signal processing systems using the PIC16C74 and provides application code modules and examples for DTMF tone generation, a 60 Hz notch filter, and a simple PID compensator for control systems. These routines can also be used with other PIC16C6X and PIC16C7XXX processors with minor modifications and the addition of external analog I/O devices.

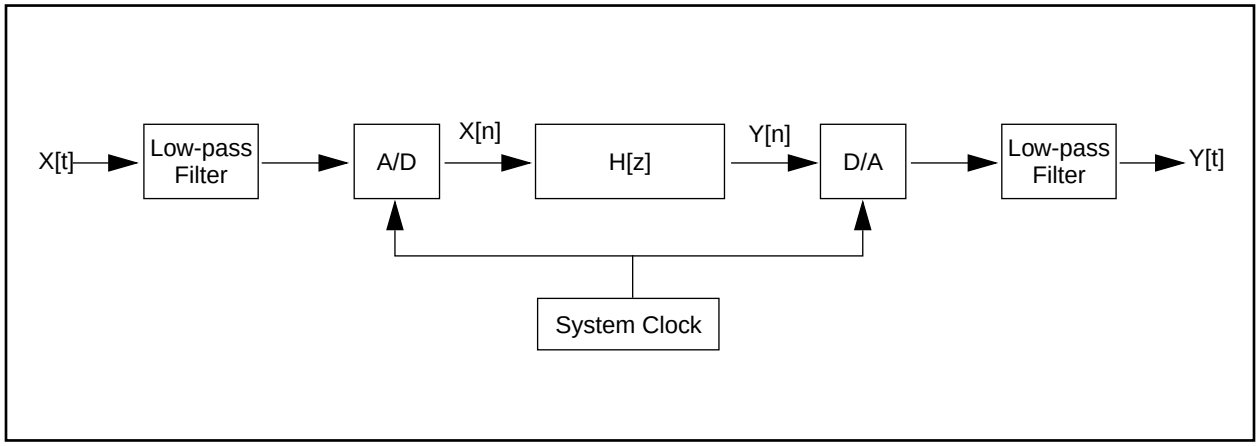
The use of general purpose microcontrollers for low-end digital signal processing applications has become more commonplace these days with the availability of higher speed processors. Since most signal processing systems consist of a host processor and dedicated DSP chip, the use of a single microcontroller to perform both these functions provides a simpler and lower cost solution. In addition, the single chip design will consume less power which is ideal for battery powered applications. The PIC16C74 with its on-chip A/D, PWM module, and fast CPU is an ideal candidate for use in these low-bandwidth signal processing applications.

A typical signal processing system includes an A/D converter, D/A converter, and CPU that performs the signal processing algorithm as shown in Figure 1.

The input signal, *x(t)*, is first passed through an input filter (commonly called the anti-aliasing filter) whose function is to bandlimit the signal to below the Nyquist rate (one half the sampling frequency) to prevent aliasing. The signal is then digitized by the A/D converter at a rate determined by the sample clock to produce *x(n)*, the discrete-time input sequence. The system transfer function, *H(z)*, is typically implemented in the time-domain using a difference equation. The output sample, *y(n)*, is then converted back into the continuous-time signal, *y(t)*, by the D/A converter and output low-pass filter.

The calculation of the output signal using a difference equation requires a multiply and accumulate (MAC) operation. This is typically a single-cycle instruction on DSP chips but can take many cycles to perform on a standard microcontroller since it must be implemented in code. Since the digitization of the signal, calculation of the output, and output to the D/A converter all must be completed within the sample clock period, the speed at which this can be done determines the maximum bandwidth that can be achieved with the system. The relatively slow speed of most microcontrollers is the major limitation when they are used in DSP applications but the PIC16C74's fast instruction execution speed (as fast as 200 ns/instruction) can provide the performance required to implement relatively low bandwidth systems. In addition, the device's on-chip A/D and PWM modules provide all the functions needed for a single chip system. Only a few external components are needed to use the PIC16C74 for tone generation, filtering of transducer signals, or low bandwidth control.

FIGURE 1: TYPICAL SIGNAL PROCESSING SYSTEM



CODE DEVELOPMENT TOOLS

The code for these applications was written using Byte Craft's MPC C compiler. The MPC compiler provides an Integrated Development Environment (IDE) and generates highly optimized code for the entire PICmicro™ family. For new PICmicro users that are familiar with C, this is an ideal way to quickly develop code for these processors. In addition, the listing files can be studied in order to learn the details of PICmicro assembly language. The modules and examples for this application note use C for the main program body and in-line assembly language for the time-critical routines. MPC provides interrupt support so that interrupt service routines (ISRs) can be easily written in either C or assembly. This feature was used to provide a timer ISR for one of the code modules. The compiler proved to be a valuable tool that allowed both high level and assembly language routines to be written and tested quickly.

In order to provide the double precision math functions required for this application note, a couple of existing math functions written for the PIC16C54 (AN525, *Programming PIC16C5X Devices on Logical Devices*) were converted for use with MPC. The double precision multiply and addition routines were modified by first changing all RAM declarations done in EQU statements to C "*unsigned char*" variable declarations. The main body of assembly language code was preceded by "#asm" and ended by "#endasm" preprocessor directives which tell the compiler where the in-line assembly code starts and ends. Finally, any macro sections and register names that are defined differently in MPC were changed.

The assembly language routines for tone generation and filtering were also written as C functions using the compiler. Assembly language routines written in this way can be called directly from other assembly language modules or called directly from C by using the label name as a C function. Source listings for all the modules and example programs can be found in the appendices at the end of this application note. These modules can be directly compiled using the MPC compiler or, alternatively, the assembly language sections can be used with MPASM with minor modifications.

Number Representation and Math Routines

One of the challenges of using any general purpose microcontroller for signal processing algorithms is in implementing the finite word-length arithmetic required to perform the calculations. As mentioned before, the speed at which the MAC operations can be performed limits the maximum achievable bandwidth of the system. Therefore, the routines that perform the multiplication and the main signal processing algorithms need to be optimized for speed in order to obtain the highest possible bandwidth when using the PIC16C74.

The selection of word size and coefficient scaling are also important factors in the successful implementation of signal processing systems. The effects of using a fixed word length to represent the signal and do calculations fall into three categories: signal quantization, round-off error, and coefficient quantization. The signal quantization due to the A/D converter and round-off error due to the finite precision arithmetic affect the overall signal-to-noise performance of the system. Scaling of the input signal should be done before the A/D converter to use the full input range and maximize the input signal-to-noise ratio. The use of double precision math for all calculations and storing intermediate results, even if the input and output signals are represented as 8-bit words, will help to reduce the round-off error noise to acceptable levels. Coefficient quantization occurs when the calculated coefficients are truncated or rounded off to fit within the given word length. This has the effect of moving the system transfer function poles and zeros which can change the system gain, critical frequencies of filters, or stability of the system. The successful implementation of these systems requires careful design and modeling of these effects using one of the many software programs that are available. The code written for this application note was first modeled using PC MATLAB before being implemented on the PIC16C74.

The algorithms in this application note are all implemented using fixed point two's compliment arithmetic. Two math libraries were used for the examples: one 8-bit signed multiply routine that was written specifically for the tone generation algorithm, and the modified double precision routines for the PIC16C54 that were used in the filtering routine. All numbers are stored in fractional two's compliment format where the MSB is the sign bit and there is an implied decimal point right after it. This is commonly referred to as Qx format where the number after the Q represents the number of fractional bits in the word. For instance, 16 bit words with the decimal point after the MSB would be referred to as Q15. This format allows numbers over the range of -1 to 0.99 to be represented and, because the magnitude of all numbers is less than or equal to one, has the advantage that there can be no overflow from a multiplication operation.

Since calculations are done using two's compliment arithmetic, values read by the PIC16C74's A/D converter need to be converted to this format. This can be easily done if the input is set up to read values in offset binary format. In this representation, the most negative input voltage is assigned to the number 0, zero volts is assigned the number 128, and the most positive voltage input is assigned 255. Since the PIC16C74 has a unipolar input A/D converter, a bipolar input signal must be scaled to be between 0 and 5V. One way to accomplish this is to use an op-amp scaling and offset circuit. The signal should be centered at 2.5V and have a peak to peak voltage swing of 4 to 4.5V. The offset binary number can be converted to two's compliment format by simply complimenting the MSB of the word. Once the signal processing calculations are completed, the number can be converted back to offset binary by complimenting the MSB before it is written to the PWM module. A similar level shifting circuit can be used at the PWM output to restore the DC level of the signal. Using this technique allows a wide range of analog input voltages to be handled by the PIC16C74.

A/D and D/A Conversion

The PIC16C74's internal 8-bit A/D converter and PWM modules can be used to implement analog I/O for the system. The A/D converter along with an external anti-aliasing filter provides the analog input for the system. Depending on the input signal bandwidth and the sampling frequency, the filter can be a simple single pole RC filter or a multiple pole active filter. The PWM output along with an external output "smoothing" filter provides the D/A output for the system. This can be a simple RC filter if the PWM frequency is much higher (five to ten times) than the analog signal that is being output. Alternatively, an active filter can also be used at the PWM output . Since the use of the A/D and PWM modules is covered in detail in the data sheet for the part, they will not be covered here. In addition, since the PIC16C74's A/D converter is similar to the PIC16C71 and the PWM module is the same as the PIC16C74, the use of these is also covered in application notes AN546, AN538, and AN539.

Appendix A contains the listing for the C module "*ANALOGIO.C*" that has the functions that read the A/D converter input, initialize the PWM module, and write 8-bit values to the PWM module. The number format (offset binary or two's compliment) for the A/D and PWM values as well as the PWM resolution and mode are set using "`#define`" pragmas at the beginning of the module. The `get_sample()` function takes the A/D input multiplexor channel number as an argument and returns the measured input value. The `init_PWM()` function takes the PWM period register PR2 value as an argument. The `write_PWM()` function takes the output values for PWM module1 and 2 and writes them to the appropriate registers using the specified resolution. If the second argument to the function is 0, the registers for PWM module 2 are unaffected. The PWM resolution is always 8-bits but the mode used depends on the PWM frequency.

The A/D conversions need to be performed at the system sample rate which requires that some form of sample clock be generated internally or input from an external source. One way to generate this clock internally, in software with minimal effort, is to use the Timer2 interrupt. Since Timer2 is used to generate the PWM period, enabling the Timer2 interrupt and using the Timer2 postscaler can generate an interrupt at periods that are integer divisors of the PWM period. The ISR can set a software "sample flag" that is checked by the main routine. Once the sample flag is asserted by the ISR, the main routine can then clear it and perform the signal processing operation, output the next sample, and then wait for the sample flag to be asserted true again. Alternatively, a separate timer/counter or external clock input can be used for the system sample clock. The latter two methods have the advantage that the PWM frequency can be set independent of the sampling period. For best results, the PWM frequency should be set for at least five times the maximum frequency of the analog signal that is bring reproduced. The example programs illustrate the use of both of the methods for generating an internal sample clock.

Tone Generation

For systems that need to provide audible feedback or to provide DTMF signaling for telcom applications, the PIC16C74's PWM module can be used to generate these signals. One way to do this is to output samples of a sinusoidal waveform to the PWM module at the system sampling rate. This method is relatively simple but is limited to single tones and may require large amounts of memory depending on the number of samples used per cycle of the waveform and the number of tones that need to be generated. A more efficient method of generating both single and dual-tone signals is to use a difference equation method. This method uses a difference equation that is derived from the z-transform of a sinusoid as follows:

The z-transform of a sinusoid is

$$\frac{z^{-1}\sin\omega T}{1 - 2z^{-1}\cos\omega T + z^{-2}}$$

where the period $\omega = 2\pi f$ and T is the sampling period.

If this is interpreted as the transfer function $H(z) = Y(z)/X(z)$ then the difference equation can be found taking the inverse z-transform and applying the associated shift theorem as follows:

rearranging:

$$Y(z)(1 - 2z^{-1}\cos\omega T + z^{-2}) = X(z)(z^{-1}\sin\omega T)$$

$$Y(z) = z^{-1}X(z)\sin\omega T + z^{-1}Y(z)2\cos\omega T - z^{-2}Y(z)$$

taking the inverse z-transform:

$$Z^{-1}[Y(z)] = Z^{-1}[z^{-1}X(z)\sin\omega T + z^{-1}Y(z)2\cos\omega T - z^{-2}Y(z)]$$

$$y(n) = \sin\omega T\, x(n - 1) + 2\cos\omega T\, y(n - 1) - y(n - 2)$$

If we let $a = \sin\omega T$ and $b = \cos\omega T$, the equation can be written as:

$$y(n) = a\, x(n - 1) + 2b\, y(n - 1) - y(n - 2)$$

thus we have a difference equation with coefficients *a* and *b*. Note that only two coefficients are needed to generate a sinusoidal output sequence. These are calculated from the relationship above and stored in memory for use by the tone generation algorithm.

If we input an impulse to this system (x(n) = 1 at n = 0 and is zero elsewhere) then the output of the system will be a discrete-time sinusoidal sequence. Note that at n = 0, the output will always be 0 and x(n) is only 1 at n = 1 so the sequence becomes:

$y(0) = 0$
$y(1) = a$
$y(n) = 2b\, y(n - 1) - y(n - 2)$
for n equal to or greater than 2

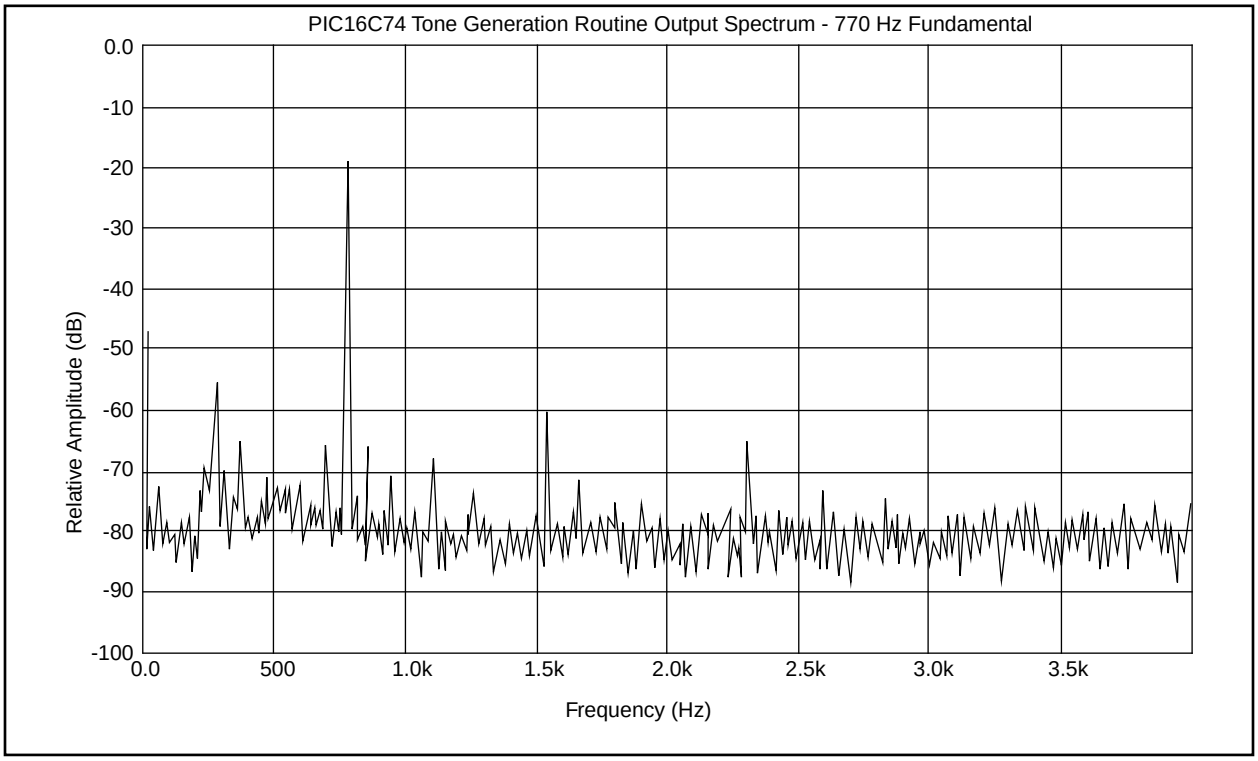
In order to further simplify the implementation of the algorithm, we can omit the first sample period. Since the output is already at 0 before starting, this will make no difference in the final output other than the fact that it will be time shifted by one sample. To generate dual tones, the algorithm is executed once for each tone and the two output samples are summed together. Since the output must be calculated and output to the D/A each sample period, a limitation exists on the frequency of the tone that can be produced for a given sample rate and processor speed. The higher the ratio of the sample clock to the tone frequency, the better, but a sample rate of at least three to four times the highest tone output should produce a sine wave with acceptable distortion.

Appendix B contains the listing for the "PICTONE.C" module which uses the difference equation method to produce variable length tones from the PWM module. Timer2 is used to generate the PWM period as well as the sample clock and tone duration timer. To send a tone, the coefficients and duration are written to the appropriate variables and then the tone routine is called. If the a2 and b2 coefficients are cleared, the routine will only generate a single tone sequence. The difference equation algorithm uses 8-bit signed math routines for the multiply operations. Using 8-bit coefficients reduces the accuracy by which the tones can be generated but greatly reduces the number of processor cycles needed to perform the algorithm since only single precision arithmetic is used. The spectrum of a single tone signal generated using this routine is shown in Figure 2.

Note that the second harmonic is better than 40 dB below the fundamental. Accuracy of this particular tone is better than 0.5%.

An example program "DTMFGEN.C" illustrates the use of the tone module to generate the 16 standard DTMF tones used for dialing on the telephone system. A sampling rate of 6.5 kHz was used which allows dual tones to be generated on a processor running at 10 MHz. Accuracy with respect to the standard DTMF frequencies is better than 1% for all tones and all harmonics above the fundamental frequency are greater than 30 dB down.

FIGURE 2: SINGLE TONE SIGNAL



Digital Filters

Digital filters with critical frequencies up to a kilohertz or so can be implemented on the PIC16C74. Digital filters fall into two classes: Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters. FIR filters require more coefficients and multiplication operations to implement practical filters and are not as well suited for implementation on the PIC16C74. IIR type filters are typically designed starting with an analog filter prototype and then performing an analog to digital transformation to produce the digital filter coefficients. The subject of digital filter design is not within the scope of this application note but there are many excellent texts that cover the theory and design of these filters.

The implementation of a second-order IIR filter is done by using a second-order difference equation. A second-order infinite impulse response (IIR) filter has a transfer function of the form:

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

Where a_1 , a_2, b_0 , b_1 , and b_2 are the coefficients of the polynomials of the system transfer function that, when factored, yield the system poles and zeros. The difference equation found by taking the inverse z-transform and applying the shift theorem is:

$$y(n) =$$

$$b_0 x(n) + b_1 x(n-1) + b_2 x(n-2) - a_1 y(n-1) - a_2 y(n-2)$$

Since the transfer function coefficients are used directly in the difference equation, this is often called the “Direct Form I” implementation of a digital filter. This form has its limitations due to numerical accuracy issues but is effective for implementing second-order systems.

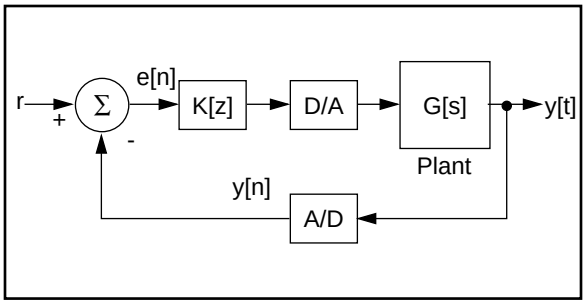
Appendix C contains the listing for the general-purpose filter routine “IIR_FILT.C” that can be used to implement low-pass, high-pass, bandpass, and bandstop (notch) filters. The filter() function takes an 8-bit input value x(n) and calculates the output value y(n) . The filter coefficients are stored as 16-bit two’s compliment numbers and computation of the output is done using double precision arithmetic. Since the coefficients generated from the filter design program will be in decimal form, they need to be scaled to be less than 1 and then multiplied by 32,768 to put them in Q15 format. Additional scaling by factors of two may be required to prevent overflow of the sum during calculations. If this is done, the output must be multiplied by this scale factor to account for this. The “IIR_FILT.C” module contains two other subroutines required for the filtering program. One if these is a decimal adjust subroutine to restore the decimal place after two 16-bit Q15 numbers are multiplied. The subroutine shifts the 32-bit result left by one to get rid of the extra sign bit. The other routine scales the output by factors of two and is used after the output of the filter has been calculated to account for the scaling of the coefficients.

An example program “NOTCH_60.C” is provided that illustrates the implementation of a 60 Hz notch filter using the “IIR_FILT.C” module. The filter was modeled and designed using PC MATLAB before being implemented on the PIC16C74. A sample rate of 1 kHz is used which means that signals up to a few hundred hertz can be processed. The filter provides an attenuation of about 40 dB at 60 Hz and can be used to remove interference from sensor signals in a system.

Digital Control

A low bandwidth digital control system can be implemented on the PIC16C74 using the analog I/O and IIR filter routines. A typical digital control system is shown below:

FIGURE 3: TYPICAL DIGITAL CONTROL SYSTEM



The input, r, is the reference input and *y(t)* is the continuous-time output of the system. *G(s)* is the analog transfer function of the plant (controlled system) and *K(z)* is the digital compensator. The error signal is calculated by subtracting the measured output signal, *y(n)*, from the reference. The controller transfer function is essentially a filter that is implemented in the time-domain using a difference equation. Since digital control system design is a complex subject and the design of a suitable compensator depends on the system being controlled and the performance specifications, only the implementation issues will be discussed.

One popular and well understood compensator is the Proportional-Integral-Derivative (PID) controller whose transfer function is of the form:

$$K(z) = K_P + \frac{K_I}{1 - z^{-1}} + K_D(1 - z^{-1})$$

Where K_P is the proportional gain, K_I is the integral gain ,and K_D is the derivative gain. The transfer function can be implemented directly or can be put in the form of a standard second-order difference equation from the modified transfer function as shown below:

$$H(z) = \frac{(K_I T^2 + K_P T + K_D) - (2K_D + K_P T)z^{-1} + K_D z^{-2}}{T(1 - z^{-1})}$$

$$y(n) = (K_P + K_I T + \frac{K_D}{T})x(n) - (K_P + \frac{2K_D}{T})x(n - 1) + \frac{K_D}{T}x(n - 2) - y(n - 1)$$

Since the numerator coefficients will be greater than one, a gain factor K needs to be factored out so that the resulting coefficients are less than one. In this way, the IIR filter routine can be used to implement the controller. After the filter routine, the output y needs to be multiplied by K before being output to the PWM module. Since the gain can be high, this result needs to be checked for overflow and limited to the maximum 8-bit value, if required. Saturating the final result prevents the system from going unstable if overflow in the math does occur. The gains can also be applied externally at the D/A output. For example, the PWM can drive a power op-amp driver that provides a ± 20 volt swing for a DC motor.

RESULTS AND CONCLUSION

The results obtained using the PIC16C74 in these applications were impressive. The tone generation routines produce very clean sinusoidal signals and DTMF tones generated using this routine have been used to dial numbers over the telephone system with excellent results. In addition, tones used for audible feedback are more pleasing to the ear than those generated from a port pin as is typically done on processors without PWM modules. Using the PIC16C74 to generate these tones eliminates the need for special DTMF generator IC's thus reducing the cost and simplifying the design. The tone routine requires approximately 125 instruction cycles to calculate an output sample for a single tone output and 230 instruction cycles to calculate an output sample for a dual tone output.

The IIR filtering routines produce good results and have been used to filter 60 Hz signals on sensor lines and also to implement a simple PID controller system with excellent results. The IIR routine takes approximately 1670 instruction cycles to calculate the output. Table 1 shows the performance that can be expected with the PIC16C74 for various processor speeds.

In conclusion, the PIC16C74 provides the necessary performance to provide these simple, low bandwidth signal processing operations. This means that products using this device can benefit from cost and power savings by eliminating specialized components that normally perform these tasks.

References

Antoniou, A. Digital Filters: Analysis and Design. NY: McGraw-Hill Book Co., 1979.

Openheim, A.V. and Schafer, R.W. Digital Signal Processing. Englewood Cliffs, N.J.: Prentice-Hall, Inc., 1975.

TABLE 1: PIC16C74 IIR FILTER PERFORMANCE

	4 MHz	8 MHz	10 MHz	16 MHz	20 MHz
A/D Input (35 cycles + 15 µs)	50 µs	32.5	29	23.75	22
IIR Filter (1850 cycles)	1850	925	740	462.5	370
PWM Output (62 cycles)	62	31	24.8	15.5	12.4
Total	1962	988.5	793.8	501.75	368.4
Max. Sampling Frequency	~500 Hz	~1000 Hz	~1250 Hz	~2000 Hz	~2500 Hz

FIGURE 4: SCHEMATIC

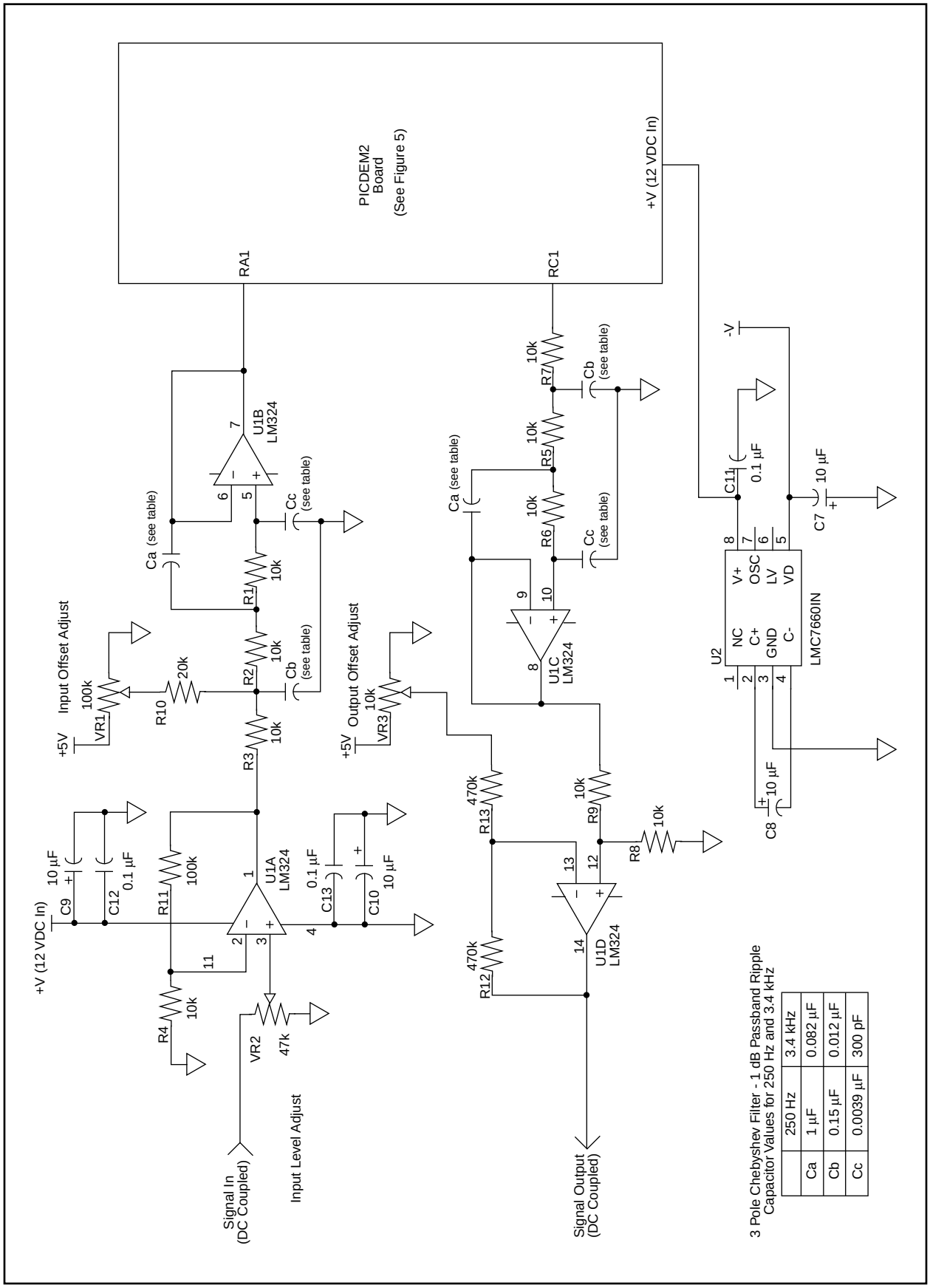
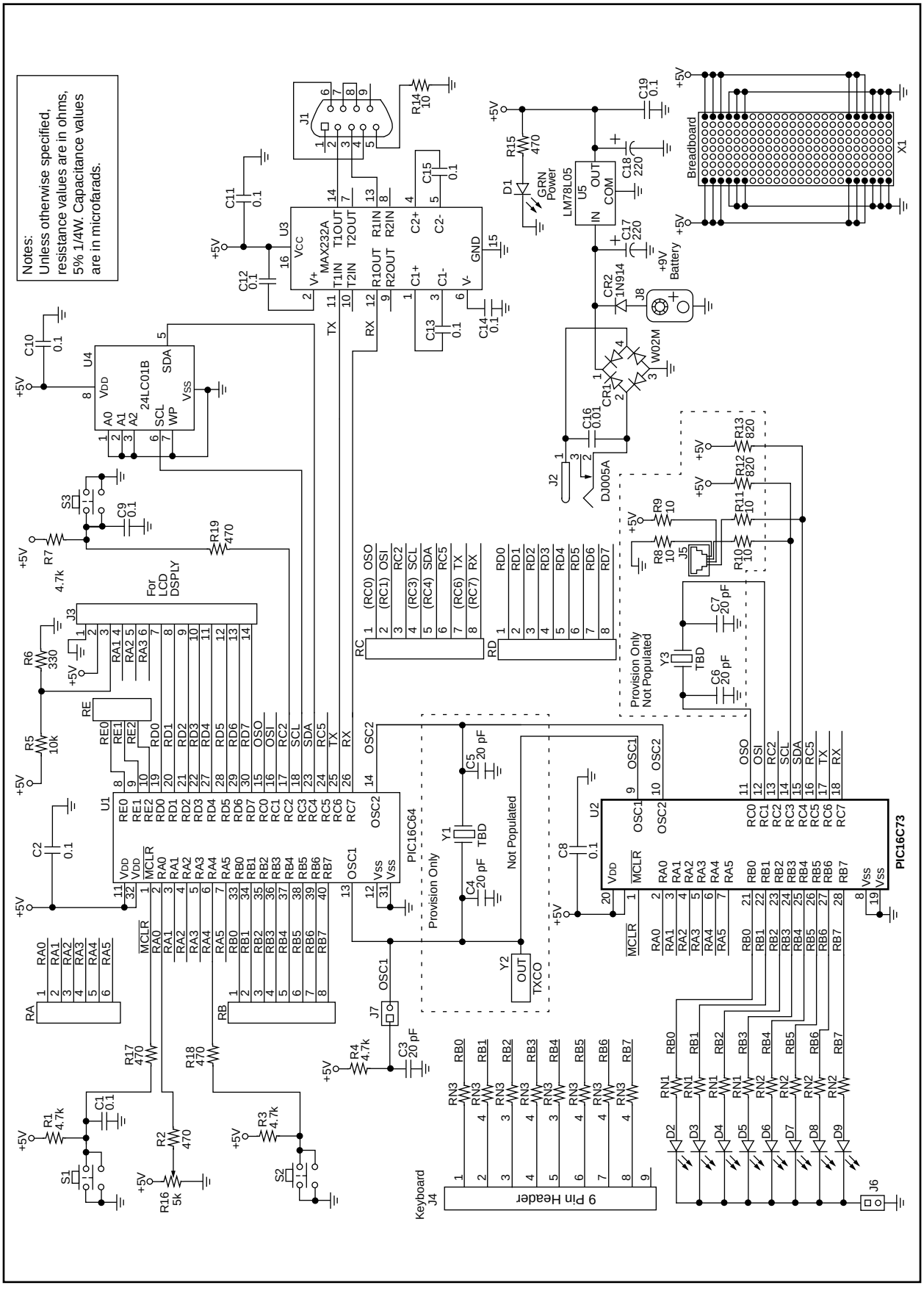


FIGURE 5: PICDEM2 SCHEMATIC TIE-IN



Please check the Microchip BBS for the latest version of the source code. Microchip's Worldwide Web Address: www.microchip.com; Bulletin Board Support: MCHIPBBS using CompuServe® (CompuServe membership not required).

APPENDIX A: ANALOG I/O MODULE

```
/************************************************************************
* Analog I/O Module
*
* Written for "Digital Signal Processing with the PIC16C74" Application Note
*
* This module contains functions that read the A-D inputs, initialize the PWM
* ports, and write values to the PWM ports.
*
* D. Mostowfi 4/95
************************************************************************/
#define active          1       /* define active as 1 */
#define LOW             0       /* define LOW as 0 */
#define HIGH            1       /* define HIGH as 1 */

#define OFFSET          0       /* define offset binary mode as 0 */
#define TWOS            1       /* define two's compliment mode as 1 */

#define AD_FORMAT       TWOS    /* define A-D format as TWOS */
#define PWM_FORMAT      TWOS    /* define PWM format as TWOS */
#define PWM_RES         HIGH    /* define PWM resolution as HIGH */

bits FLAGS;                     /* general purpose flags */
#define sample_flag     FLAGS.1 /* define sample_flag as FLAGS.1 */


/************************************************************************
* A-D Converter Routine - reads A-D converter inputs
*
* usage:
*       - call get_sample(channel #)
*       - returns 8 bit value
************************************************************************/
char get_sample(char channel)
{
char i;

    ADRES=0;                    /* clear ADRES */
    STATUS.C=0;                 /* clear carry */
    RLCF(channel);              /* and rotate channel 3 times */
    RLCF(channel);              /* to put in proper position */
    RLCF(channel);              /* for write to ADCON0 */
    ADCON0=channel;             /* write channel to ADCON0 */
    ADCON0.0=1;                 /* turn on A-D */
    i=0;                        /* set delay loop variable to 0 */
    while(i++<=5){};            /* delay (to ensure min sampling time) */
    ADCON0.2=1;                 /* start conversion */
    while(ADCON0.2){}           /* wait for eoc */
    ADCON0.0=0;                 /* turn off a-d converter */
    if(AD_FORMAT==TWOS){        /* if format is two's compliment */
      ADRES.7=!ADRES.7;         /* compliment MSB */
    }
    return ADRES;               /* return value in a-d result reg */
}


/************************************************************************
* PWM Initialization Routine - sets up PR2, sets output to mid-point, and
* starts timer 2 with interrupts disabled.
*
* usage:
```

```
*       - call init_PWM(PR2 register value)
*************************************************************************/
void init_PWM(char _pr2)
{
    PR2=_pr2;                   /* reload value for 40khz PWM period */
    CCP1CON.5=0;                /* set CCPxCON = 0 for 50% output */
    CCP1CON.4=0;
    CCP2CON.5=0;
    CCP2CON.4=0;
    if(PWM_RES==HIGH){          /* if resolution is high, set CCPRxH=0 and */
      CCPR1H=0x00;              /* CCPRxL=0x20 for 50% PWM duty cycle */
      CCPR1L=0x20;
      CCPR2H=0x00;
      CCPR2L=0x20;
    }
    else{
      CCPR1H=0x00;              /* if resolution is low, set CCPRxH=0 and */
      CCPR1L=0x80;              /* CCPRxL=0x80 for 50% PWM duty cycle */
      CCPR2H=0x00;
      CCPR2L=0x80;
    }
    T2CON.TMR2ON=1;             /* start timer 2 */
    PIE1.TMR2IE=0;              /* and disable timer 2 interrupt */

}


/*************************************************************************
* PWM Output Routine - writes output values to PWM ports
*
* Both high resolution and low resolution modes write 8 bit values - use of
* high or low resolution depends on PWM output period.
*
* usage:
*       - call write_PWM(channel 1 value, channel 2 value)
*         if channel 2 value=0, PWM port 2 not written
*************************************************************************/
void write_PWM(bits pwm_out1, bits pwm_out2)
{

    if(PWM_FORMAT==TWOS){       /* if format is two's compliment */
      pwm_out1.7=!pwm_out1.7;   /* compliment msb's */
      pwm_out2.7=!pwm_out1.7;
    }
    if(PWM_RES==HIGH){          /* if resolution is high */
      STATUS.C=0;               /* clear carry */
      pwm_out1=RRCF(pwm_out1);  /* rotate right and write two lsb's */
      CCP1CON.4=STATUS.C;       /* to CCP1CON4 and CCP1CON5 */
      STATUS.C=0;
      pwm_out1=RRCF(pwm_out1);
      CCP1CON.5=STATUS.C;
      if(pwm_out2!=0){          /* if pwm_out2 not 0, do the same */
        STATUS.C=0;             /* for channel 2 */
        pwm_out2=RRCF(pwm_out2);
        CCP2CON.4=STATUS.C;
        STATUS.C=0;
        pwm_out2=RRCF(pwm_out2);
        CCP2CON.5=STATUS.C;
      }
    }
    CCPR1L=pwm_out1;            /* write value to CCPR1L */
    if(pwm_out2!=0){            /* if pwm_out2 not 0, do the same */
      CCPR2L=pwm_out2;          /* for CCPR2L */
    }
}                               /* done */
```

Please check the Microchip BBS for the latest version of the source code. Microchip's Worldwide Web Address: www.microchip.com; Bulletin Board Support: MCHIPBBS using CompuServe® (CompuServe membership not required).

APPENDIX B: TONE GENERATION MODULE

```
/************************************************************************
* Tone Generation Module
*
* Written for "Digital Signal Processing with the PIC16C74" Application Note.
*
* This module contains a C callable module that generates single or dual
* tones using a difference equation method:
*
* y1(n)=a1*x(n-1)+b1*y1(n-1)-y1(n-2)
* y2(n)=a2*x(n-1)+b2*y2(n-1)-y2(n-2)
*
* The routine is written in assembly language and uses the optimized signed
* 8x8 multiply routine and scaling routine in the file 8BITMATH.C.
*
* D. Mostowfi 2/95
************************************************************************/
#include "\mpc\apnotes\8bitmath.c"  /* 8 bit signed math routines */

#define sample_flag FLAGS.1         /* sample flag */
#define no_tone2 FLAGS,2            /* no tone 2 flag */

extern char ms_cntr;                /* millisecond counter for tone loop */

char a1;                            /* first tone (low-group) coeeficient 1 */
char a2;                            /* first tone (low-group) coefficient 2 */
char b1;                            /* second tone (high group) coefficient 1 */
char b2;                            /* second tone (high group) coefficient 2 */
char duration;                      /* tone duration */

char y1;                            /* output sample y1(n) for tone 1 */
char y2;                            /* output sample y2(n) for tone 2 */


/************************************************************************
* Tone function - generates single or dual tone signals out PWM port 1.
*
* usage:
*       - write coefficients for tone 1 to a1 and b1
*       - write coefficents for tone 2 to a2 and b2 (0 if no tone 2)
*       - write duration of tone in milliseconds to duration
*       - call tone() function
************************************************************************/
void tone(void)
{

char y1_1;                          /* y1(n-1) */
char y1_2;                          /* y1(n-2) */
char y2_1;                          /* y2(n-1) */
char y2_2;                          /* y2(n-2) */

    PIR1.TMR2IF=0;                  /* clear timer 2 interrupt flag */
    PIE1.TMR2IE=1;                  /* and enable timer 2 interrupt */
    ms_cntr=0;                      /* clear ms counter */
    STATUS.RP0=0;                   /* set proper bank!!! */

#asm
        clrf    y1                  ; clear output byte and taps
        clrf    y2                  ;
        clrf    y1_1                ;
        clrf    y1_2                ;
```

```
        clrf    y2_1            ;
        clrf    y2_2            ;

        bcf     no_tone2        ; clear no tone 2 flag
        clrf    ms_cntr         ; clear millisecond counter

first_sample:
        movf    a1,W            ; first iteration
        movwf   y1              ; y1(n)=a1
        movwf   y1_1            ;
        movlw   0x00            ;
        iorwf   a2,W            ;
        btfsc   STATUS,Z        ; generate second tone (a2 !=0) ?
        bsf     no_tone2        ;
        movf    a2,W            ; y2(n)=a2
        movwf   y2              ;
        movwf   y2_1            ;
        movf    y2,W            ;
        addwf   y1,F            ; y1(n)=y1(n)+y2(n) (sum two tone outputs)

tone_loop:
        movf    ms_cntr,W       ; test to see if ms=duration (done?)
        subwf   duration,W      ;
        btfsc   STATUS,Z        ;
        goto    tone_done       ;

wait_PWM:
        btfss   FLAGS,1         ; test sample flag (sample period elapsed?)
        goto    wait_PWM        ; loop if not

        bcf     FLAGS,1         ; if set, clear sample flag

#endasm
    write_PWM((char)y1,0);      /* write y1 to PWM port */
#asm

next_sample:
        movf    b1,W            ; y1(n)=b1*y1(n-1)-y1(n-2)
        movwf   multcnd         ;
        movf    y1_1,W          ;
        movwf   multplr         ;
        call    _8x8smul        ;
        call    scale_16        ;
        movf    y1_2,W          ;
        subwf   result_l,W      ;
        movwf   y1              ;
        movf    y1_1,W          ; y1(n-2)=y1(n-1)
        movwf   y1_2            ;
        movf    y1,W            ; y1(n-1)=y1(n)
        movwf   y1_1            ;
        btfsc   no_tone2        ;
        goto    tone_loop       ;
        movf    b2,W            ; y2(n)=b2*y2(n-1)-y2(n-2)
        movwf   multcnd         ;
        movf    y2_1,W          ;
        movwf   multplr         ;
        call    _8x8smul        ;
        call    scale_16        ;
        movf    y2_2,W          ;
        subwf   result_l,W      ;
        movwf   y2              ;
        movf    y2_1,W          ; y2(n-2)=y2(n-1)
        movwf   y2_2            ;
        movf    y2,W            ; y2(n-1)=y2(n)
        movwf   y2_1            ;
```

```
        movf    y2,W            ;
        addwf   y1,F            ; y1(n)=y1(n)+y2(n) (sum two tone outputs)

        goto    tone_loop       ; go and calculate next sample

tone_done:

#endasm

    CCP1CON.5=0;                /* reset PWM outputs to mid value */
    CCP1CON.4=0;
    CCP2CON.5=0;
    CCP2CON.4=0;
    CCPR1H=0x00;
    CCPR1L=0x20;
    CCPR2H=0x00;
    CCPR2L=0x20;

    PIE1.TMR2IE=0;              /* disable timer 2 interrupts */
    PIR1.TMR2IF=0;              /* and clear timer 2 interrupt flag */

}
```

Please check the Microchip BBS for the latest version of the source code. Microchip's Worldwide Web Address: www.microchip.com; Bulletin Board Support: MCHIPBBS using CompuServe® (CompuServe membership not required).

APPENDIX C: DTMF TONE GENERATION

```
/************************************************************************
* DTMF tone generation using PIC16C74
*
* Written for the "Digital Signal Processing Using the PIC16C74" Ap Note
*
* Generates 16 DTMF tones (1-9,0,*,#,A,B,C,D) out PWM port 1
*
* Uses PICTONE.C and ANALOGIO.C modules
*
* D. Mostowfi 4/95
************************************************************************/
#include "\mpc\include\delay14.h"
#include "\mpc\include\16c74.h"      /* c74 header file */
#include "\mpc\math.h"


#include "\mpc\apnotes\analogio.c"   /* analog I/O module */
#include "\mpc\apnotes\pictone.c"    /* tone generation module */

bits pwm1;

/* Function Prototypes */

void main_isr();
void timer2_isr();

/* 16C74 I/O port bit declarations */


/* global program variables */

char tmr2_cntr;                    /* timer 2 interrupt counter */
char delay_cntr;                   /* delay time counter (10ms ticks)*/


/* Tone Coefficients for DTMF Tones */

const DTMF_1[4]={30, 51, 48, 27};
const DTMF_2[4]={30, 51, 56, 19};
const DTMF_3[4]={30, 51, 64, 11};
const DTMF_4[4]={33, 48, 48, 27};
const DTMF_5[4]={33, 48, 56, 19};
const DTMF_6[4]={33, 48, 64, 11};
const DTMF_7[4]={36, 45, 48, 27};
const DTMF_8[4]={36, 45, 56, 19};
const DTMF_9[4]={36, 45, 64, 11};
const DTMF_0[4]={40, 41, 56, 19};
const DTMF_star[4]={40, 41, 48, 27};
const DTMF_pound[4]={40, 41, 64, 11};
const DTMF_A[4]={30, 51, 75, 2};
const DTMF_B[4]={33, 48, 75, 2};
const DTMF_C[4]={36, 45, 75, 2};
const DTMF_D[4]={40, 41, 75, 2};


/************************************************************************
* main isr - 16C74 vectors to 0004h (MPC __INT() function) on any interrupt *
* assembly language routine saves W and Status registers then tests flags in
* INTCON to determine source of interrupt. Routine then calls appropriate isr.
* Restores W and status registers when done.
```

```
*************************************************************************/
void __INT(void)
{

  if(PIR1.TMR2IF){              /* timer 2 interrupt ? */
    PIR1.TMR2IF=0;              /* clear interrupt flag */
    timer2_isr();               /* and call timer 2 isr */
  }

/* Restore W, WImage, and STATUS registers */

#asm
    BCF     STATUS,RP0              ;Bank 0
    MOVF    temp_PCLATH, W
    MOVWF   PCLATH                  ;PCLATH restored
    MOVF    temp_WImage, W
    MOVWF   __WImage                ;__WImage restored
    MOVF    temp_FSR, W
    MOVWF   FSR                     ;FSR restored
    SWAPF   temp_STATUS,W
    MOVWF   STATUS                  ;STATUS restored
    SWAPF   temp_WREG,F
    SWAPF   temp_WREG,W             ;W restored
#endasm

}

/*************************************************************************
* timer 2 isr - provides PWM sample clock generation and millisecond counter
* for tone routine
*************************************************************************/
void timer2_isr(void)
{
     sample_flag=active;        /* set sample flag (150us clock) */
     PORTB.7=!PORTB.7;          /* toggle PORTB.7 at sample rate */
     if(tmr2_cntr++==7){        /* check counter */
       tmr2_cntr=0;             /* reset if max */
       ms_cntr++;               /* and increment millisecond ticks */
     }
}


void main()
{

/* initialize OPTION register */
   OPTION=0b11001111;

/* initialize INTCON register (keep GIE inactive!) */
   INTCON=0b00000000;           /* disable all interrupts */

/* initialize PIE1 and PIE2 registers (peripheral interrupts) */
   PIE1=0b00000000;             /* disable all interrupts */
   PIE2=0b00000000;

/* initialize T1CON and T2CON registers */
   T1CON=0b00000000;            /* T1 not used  */
   T2CON=0b00101000;            /* T2 postscaler=5  */

/* initialize CCPxCON registers */
   CCP1CON=0b00001100;          /* set CCP1CON for PWM mode */
   CCP2CON=0b00001100;          /* set CCP2CON for PWM mode (not used in demo) */

/* initialize SSPCON register */
   SSPCON=0b00000000;           /* serial port - not used */
```

```
/* initialize ADCONx registers */
   ADCON0=0b00000000;       /* A-D converter */
   ADCON1=0b00000010;

/* initialize TRISx register (port pins as inputs or outputs) */
   TRISA=0b00001111;
   TRISB=0b00000000;
   TRISC=0b10000000;
   TRISD=0b00001111;
   TRISE=0b00000000;

/* clear watchdog timer (not used) */
   CLRWDT();

/* initialize program variables */
   tmr2_cntr=0;

/* initialize program bit variables */
   FLAGS=0b00000000;

/* intialize output port pins (display LED's on demo board) */
   PORTB=0;

/* enable interrupts...  */

   INTCON.ADIE=1;           /* Peripheral interrupt enable */
   INTCON.GIE=1;            /* global interrupt enable */




   init_PWM(0x3e);          /* initialize PWM port */

   PORTB=0x01;              /* write a 1 to PORTB */
   a1=DTMF_1[0];            /* and send a DTMF "1" */
   b1=DTMF_1[1];
   a2=DTMF_1[2];
   b2=DTMF_1[3];
   duration=150;
   tone();
   Delay_Ms_20MHz(200);     /* delay 100ms (200/2 using MPC delays) */


   PORTB=0x02;              /* write a 2 to PORT B */
   a1=DTMF_2[0];            /* and send a DTMF "2" */
   b1=DTMF_2[1];
   a2=DTMF_2[2];
   b2=DTMF_2[3];
   duration=150;
   tone();
   Delay_Ms_20MHz(200);     /* delay 100ms (200/2 using MPC delays) */


   PORTB=0x03;              /* write a 3 to PORTB */
   a1=DTMF_3[0];            /* and send a DTMF "3" */
   b1=DTMF_3[1];
   a2=DTMF_3[2];
   b2=DTMF_3[3];
   duration=150;
   tone();
   Delay_Ms_20MHz(200);     /* delay 100ms (200/2 using MPC delays) */


   PORTB=0x04;              /* write a 4 to PORTB */
   a1=DTMF_4[0];            /* and send a DTMF "4" */
   b1=DTMF_4[1];
```

```
a2=DTMF_4[2];
b2=DTMF_4[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */
PORTB=0x05;           /* write a 5 to PORTB */
a1=DTMF_5[0];         /* and send a DTMF “5” */
b1=DTMF_5[1];
a2=DTMF_5[2];
b2=DTMF_5[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


PORTB=0x06;           /* write a 6 to PORTB */
a1=DTMF_6[0];         /* and send a DTMF “6” */
b1=DTMF_6[1];
a2=DTMF_6[2];
b2=DTMF_6[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


PORTB=0x07;           /* write a 7 to PORTB */
a1=DTMF_7[0];         /* and send a DTMF “7” */
b1=DTMF_7[1];
a2=DTMF_7[2];
b2=DTMF_7[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


PORTB=0x08;           /* write a 8 to PORTB */
a1=DTMF_8[0];         /* and send a DTMF “8” */
b1=DTMF_8[1];
a2=DTMF_8[2];
b2=DTMF_8[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


PORTB=0x09;           /* write a 9 to PORTB */
a1=DTMF_9[0];         /* and send a DTMF “9” */
b1=DTMF_9[1];
a2=DTMF_9[2];
b2=DTMF_9[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


PORTB=0x0;            /* write a 0 to PORTB */
a1=DTMF_0[0];         /* and send a DTMF “0” */
b1=DTMF_0[1];
a2=DTMF_0[2];
b2=DTMF_0[3];
duration=150;
tone();
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */
Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */
```

```
   PORTB=0x0e;           /* write a 0x0e to PORTB */
   a1=DTMF_star[0];      /* and send a DTMF "*" */
   b1=DTMF_star[1];
   a2=DTMF_star[2];
   b2=DTMF_star[3];
   duration=250;
   tone();
   Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


   PORTB=0x0f;           /* write a 0x0f to PORTB */
   a1=DTMF_pound[0];     /* and send a DTMF "#" */
   b1=DTMF_pound[1];
   a2=DTMF_pound[2];
   b2=DTMF_pound[3];
   duration=250;
   tone();
   Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */
   Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */



   PORTB=0x0a;           /* write a 0x0a to PORTB */
   a1=DTMF_A[0];         /* and send a DTMF "A" */
   b1=DTMF_A[1];
   a2=DTMF_A[2];
   b2=DTMF_A[3];
   duration=250;
   tone();
   Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


   PORTB=0x0b;           /* write a 0x0b to PORTB */
   a1=DTMF_B[0];         /* and send a DTMF "B" */
   b1=DTMF_B[1];
   a2=DTMF_B[2];
   b2=DTMF_B[3];
   duration=250;
   tone();
   Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */


   PORTB=0x0c;           /* write a 0x0c to PORTB */
   a1=DTMF_C[0];         /* and send a DTMF "C" */
   b1=DTMF_C[1];
   a2=DTMF_C[2];
   b2=DTMF_C[3];
   duration=250;
   tone();
   Delay_Ms_20MHz(200);  /* delay 100ms (200/2 using MPC delays) */



   PORTB=0x0d;           /* write a 0x0d to PORTB */
   a1=DTMF_D[0];         /* and send a DTMF "D" */
   b1=DTMF_D[1];
   a2=DTMF_D[2];
   b2=DTMF_D[3];
   duration=250;
   tone();

   PORTB=0;              /* write a 0 to PORTB */

   while(1){}            /* done (loop) */

}
```

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APPENDIX D: IIR FILTER MODULE

```
/************************************************************************
* Second-Order IIR Filter Module
*
* Written for "Digital Signal Processing with the PIC16C74" Application Note.
*
* This routine implements an IIR filter using a second order difference
* equation of the form:
*
* y(n) = b0*x(n)+b1*x(n-1)+b2*x(n-2)+a1*y(n-1)+a2*y(n-2)
*
* D. Mostowfi 3/95
************************************************************************/
#include "\mpc\apnotes\dbl_math.c"

bits            x_n;            /* input sample x(n) */
unsigned long   y_n;            /* output sample y(n) */
unsigned long   x_n_1;          /* x(n-1) */
unsigned long   x_n_2;          /* x(n-2) */
unsigned long   y_n_1;          /* y(n-1) */
unsigned long   y_n_2;          /* y(n-2) */

char            rmndr_h;        /* high byte of remainder from multiplies */
char            rmndr_l;        /* low byte of remainder from multiplies */

#define A1_H    0xd2            /* filter coefficients */
#define A1_L    0x08            /* for 60Hz notch filter */
#define A2_H    0x11            /* Fs= 1kHz */
#define A2_L    0x71
#define B0_H    0x18
#define B0_L    0xbb
#define B1_H    0xd2
#define B1_L    0x08
#define B2_H    0x18
#define B2_L    0xb9


/************************************************************************
* Filter initialization - clears all taps in memory.
*
* usage:
*       - call init_filter()
*         use at program initialization
************************************************************************/
void init_filter(){

#asm

        clrf      y_n           ; clear output value
        clrf      y_n+1         ;
        clrf      y_n_1         ; and all filter "taps"
        clrf      y_n_1+1       ;
        clrf      y_n_2         ;
        clrf      y_n_2+1       ;
        clrf      x_n_1         ;
        clrf      x_n_1+1       ;
        clrf      x_n_2         ;
        clrf      x_n_2+1       ;

#endasm
```

```
}


/*****************************************************************************
* Assembly language subroutines for main filter() function
*****************************************************************************/
#asm

;
; Add Remainder subroutine - adds remainder from multiplies to ACCc
;

add_rmndr:
        btfss     sign,7          ; check if number is negative
        goto      add_r_start     ; go to add_r_start if not
        comf      ACCcLO          ; if so, negate number in ACC
        incf      ACCcLO          ;
        btfsc     STATUS,Z        ;
        decf      ACCcHI          ;
        comf      ACCcHI          ;
        btfsc     STATUS,Z        ;
        comf      ACCbLO          ;
        incf      ACCbLO          ;
        btfsc     STATUS,Z        ;
        decf      ACCbHI          ;
        comf      ACCbHI          ;

add_r_start:
        movf      rmndr_l,W       ;  get low byte of remainder
        addwf     ACCcLO          ;  and add to ACCcLO
        btfsc     STATUS,C        ;  check for overflow
        incf      ACCcHI          ;  if overflow, increment ACCcHI
        movf      rmndr_h,W       ;  get high byte of remainder
        addwf     ACCcHI          ;  and add to ACCcHI
        btfsc     STATUS,C        ;  check for overflow
        incf      ACCbLO          ;  if overflow, increment ACCbLO

        btfss     sign,7          ; check if result negative
        goto      add_r_done      ; if not, go to add_r_done
        comf      ACCcLO          ; if so, negate result
        incf      ACCcLO          ;
        btfsc     STATUS,Z        ;
        decf      ACCcHI          ;
        comf      ACCcHI          ;
        btfsc     STATUS,Z        ;
        comf      ACCbLO          ;
        incf      ACCbLO          ;
        btfsc     STATUS,Z        ;
        decf      ACCbHI          ;
        comf      ACCbHI          ;

add_r_done:
        retlw     0               ; done

;
; Decimal Adjust Subroutine - used after each Q15 multiply to convert Q30 result
; to Q15 number

dec_adjust:
        bcf       sign,7          ; clear sign
        btfss     ACCbHI,7        ; test if number is negative
        goto      adjust          ; go to adjust if not
        bsf       sign,7          ; set sign if negative

        comf      ACCcLO          ; and negate number
        incf      ACCcLO
```

```
        btfsc     STATUS,Z
        decf      ACCcHI
        comf      ACCcHI
        btfsc     STATUS,Z
        comf      ACCbLO
        incf      ACCbLO
        btfsc     STATUS,Z
        decf      ACCbHI
        comf      ACCbHI

adjust:
        rlf       ACCcHI          ; rotate ACC left 1 bit
        rlf       ACCbLO          ;
        rlf       ACCbHI          ;

        btfss     sign,7          ; check if result should be negative
        goto      adj_done        ; if not, done
        comf      ACCbLO          ; if result negative, negate ACC
        incf      ACCbLO
        btfsc     STATUS,Z
        decf      ACCbHI
        comf      ACCbHI

adj_done:
        retlw     0               ; done

;
; Output Scaling Routine - used to scale output samples by factors of
; 2, 4, or 8 at end of filter routine
;
scale_y_n:
        bcf       sign,7          ; clear sign,7
        btfss     y_n+1,7         ; test if y(n) negative
        goto      start_scale     ; go to start_scale if not
        bsf       sign,7          ; set sign,7 if negative
        comf      y_n             ; and compliment y(n)
        incf      y_n             ;
        btfsc     STATUS,Z        ;
        decf      y_n+1           ;
        comf      y_n+1           ;

start_scale:
        bcf       STATUS,C        ; clear carry
        rlf       y_n+1           ; and rotate y(n) left
        rlf       y_n             ;
        bcf       STATUS,C        ;
        rlf       y_n+1           ;
        rlf       y_n             ;
        bcf       STATUS,C        ;
        rlf       y_n+1           ;
        rlf       y_n             ;

        btfss     sign,7          ; test if result is negative
        goto      scale_y_done    ; go to scale_y_done if not
        comf      y_n             ; negate y(n) if result is negative
        incf      y_n             ;
        btfsc     STATUS,Z        ;
        decf      y_n+1           ;
        comf      y_n+1           ;

scale_y_done:
        retlw    0                ; done


#endasm
```

```
/*************************************************************************
* Filter function - filter takes current input sample, x(n), and outputs next
* output sample, y(n).
*
* usage:
*       - write sample to be filtered to x_n
*       - call filter()
*       - output is in MSB of y_n (y_n=MSB, y_n+1=LSB)
*
*************************************************************************/
void filter(){

#asm

        clrf    y_n             ; clear y(n) before starting
        clrf    y_n+1           ;

        clrf    ACCbLO          ; move x(n) to ACCbHI
        movf    x_n,W           ; (scale 8 bit - 16 bit input)
        movwf   ACCbHI          ;

        movlw   B0_H            ; get coefficient b0
        movwf   ACCaHI          ; y(n)=b0*x(n)
        movlw   B0_L            ;
        movwf   ACCaLO          ;
        call    D_mpyF          ;
        movf    ACCcHI,W        ; save remainder from multiply
        movwf   rmndr_h         ;
        movf    ACCcLO,W        ;
        movwf   rmndr_l         ;
        call    dec_adjust      ;
        movf    ACCbHI,W        ;
        movwf   y_n+1           ;
        movf    ACCbLO,W        ;
        movwf   y_n             ;

        movlw   B1_H            ; get coefficient b1
        movwf   ACCaHI          ; y(n)=y(n)+b1*x(n-1)
        movlw   B1_L            ;
        movwf   ACCaLO          ;
        movf    x_n_1+1,W       ;
        movwf   ACCbHI          ;
        movf    x_n_1,W         ;
        movwf   ACCbLO          ;
        call    D_mpyF          ;
        call    add_rmndr       ; add in remainder from previous multiply
        movf    ACCcHI,W        ; and save new remainder
        movwf   rmndr_h         ;
        movf    ACCcLO,W        ;
        movwf   rmndr_l         ;
        call    dec_adjust      ;
        movf    y_n+1,W         ;
        movwf   ACCaHI          ;
        movf    y_n,W           ;
        movwf   ACCaLO          ;
        call    D_add           ;
        movf    ACCbHI,W        ;
        movwf   y_n+1           ;
        movf    ACCbLO,W        ;
        movwf   y_n             ;

        movlw   B2_H            ; get coefficient b2
        movwf   ACCaHI          ; y(n)=y(n)+b2*x(n-2)
        movlw   B2_L            ;
        movwf   ACCaLO          ;
        movf    x_n_2+1,W       ;
```

```
        movwf   ACCbHI          ;
        movf    x_n_2,W         ;
        movwf   ACCbLO          ;
        call    D_mpyF          ;
        call    add_rmndr       ; add in remainder from previous multiply
        movf    ACCcHI,W        ; and save new remainder
        movwf   rmndr_h         ;
        movf    ACCcLO,W        ;
        movwf   rmndr_l         ;
        call    dec_adjust      ;
        movf    y_n+1,W         ;
        movwf   ACCaHI          ;
        movf    y_n,W           ;
        movwf   ACCaLO          ;
        call    D_add           ;
        movf    ACCbHI,W        ;
        movwf   y_n+1           ;
        movf    ACCbLO,W        ;
        movwf   y_n             ;

        movlw   A1_H            ; get coefficient a1
        movwf   ACCaHI          ; y(n)=y(n)+a1*y(n-1)
        movlw   A1_L            ;
        movwf   ACCaLO          ;
        movf    y_n_1+1,W       ;
        movwf   ACCbHI          ;
        movf    y_n_1,W         ;
        movwf   ACCbLO          ;
        call    D_mpyF          ;
        call    add_rmndr       ; add in remainder from previous multiply
        movf    ACCcHI,W        ; and save new remainder
        movwf   rmndr_h         ;
        movf    ACCcLO,W        ;
        movwf   rmndr_l         ;
        call    dec_adjust      ;
        movf    y_n+1,W         ;
        movwf   ACCaHI          ;
        movf    y_n,W           ;
        movwf   ACCaLO          ;
        call    D_sub           ;
        movf    ACCbHI,W        ;
        movwf   y_n+1           ;
        movf    ACCbLO,W        ;
        movwf   y_n             ;

        movlw   A2_H            ; get coefficient a2
        movwf   ACCaHI          ; y(n)=y(n)+a2*y(n-2)
        movlw   A2_L            ;
        movwf   ACCaLO          ;
        movf    y_n_2+1,W       ;
        movwf   ACCbHI          ;
        movf    y_n_2,W         ;
        movwf   ACCbLO          ;
        call    D_mpyF          ;
        call    add_rmndr       ;
        call    dec_adjust      ;
        movf    y_n+1,W         ;
        movwf   ACCaHI          ;
        movf    y_n,W           ;
        movwf   ACCaLO          ;
        call    D_sub           ;
        movf    ACCbHI,W        ;
        movwf   y_n+1           ;
        movf    ACCbLO,W        ;
        movwf   y_n             ;
```

```
        movf    x_n_1,W         ; x(n-2)=x(n-1)
        movwf   x_n_2           ;
        movf    x_n_1+1,W       ;
        movwf   x_n_2+1         ;
        movf    x_n,W           ; x(n-1)=x(n)
        movwf   x_n_1+1         ;
        clrf    x_n_1           ;

        movf    y_n_1,W         ; y(n-2)=y(n-1)
        movwf   y_n_2           ;
        movf    y_n_1+1,W       ;
        movwf   y_n_2+1         ;
        movf    y_n,W           ; y(n-1)=y(n)
        movwf   y_n_1           ;
        movf    y_n+1,W         ;
        movwf   y_n_1+1         ;

        call    scale_y_n       ;

        movf    y_n+1,W         ; shift lsb of y_n to msb
        movwf   y_n             ;

#endasm
}
```

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APPENDIX E: NOTCH FILTER

```
/************************************************************************
* 60 Hertz Notch Filter
*
* Written for "Digital Signal Processing with the PIC16C74" Application Note.
*
* This example program use the filter() function to implement a 60Hz notch
* filter. T0 is used to generate a 1kHz sample clock. The program samples the
* input signal x(n) on A-D channel 1, calls the filter routine  signal, and
* outputs y(n) to PWM channel 1.
*
* If FILTER set to 0, performs straight talkthru from A-D to PWM output.
* T0 period can be changed to cary the sample rate.
*
* D. Mostowfi 4/95
************************************************************************/
#include "\mpc\include\16c74.h"         /* c74 header file */

#include "\mpc\apnotes\analogio.c"      /* analog I/O module */
#include "\mpc\apnotes\iir_filt.c"      /* iir filter module */

#define FILTER          1

/* Function Prototypes */
void main_isr();
void timer0_isr();


/************************************************************************
* main isr - 16C74 vectors to 0004h (MPC __INT() function) on any interrupt *
* assembly language routine saves W and Status registers then tests flags in
* INTCON to determine source of interrupt. Routine then calls appropriate isr.
* Restores W and status registers when done.
************************************************************************/
void __INT(void)
{

    if(INTCON.T0IF){                    /* timer 0 interrupt ? */
      INTCON.T0IF=0;                    /* clear interrupt flag */
      timer0_isr();                     /* and call timer 0 isr */
    }

/* Restore W, WImage, and STATUS registers */

#asm
        BCF     STATUS,RP0              ;Bank 0
        MOVF    temp_PCLATH, W
        MOVWF   PCLATH                  ;PCLATH restored
        MOVF    temp_WImage, W
        MOVWF   __WImage                ;WImage restored
        MOVF    temp_FSR, W
        MOVWF   FSR                     ;FSR restored
        SWAPF   temp_STATUS,W
        MOVWF   STATUS                  ;RP0 restored
        SWAPF   temp_WREG,F
        SWAPF   temp_WREG,W             ;W restored
#endasm

}

/************************************************************************
* timer 0 interrupt service routine
************************************************************************/
void timer0_isr(void)
```

```
{
    TMR0=100;                   /* reload value for 1ms period */
    PORTB.0=!PORTB.0;           /* toggle PORTB.0 */
    sample_flag=active;         /* set sample flag */
}


void main()
{

/* initialize OPTION register */
    OPTION=0b00000011;          /* assign prescaler to T0 */

/* initialize INTCON register (keep GIE inactive!) */
    INTCON=0b00000000;          /* disable all interrupts */

/* initialize PIE1 and PIE2 registers (periphreal interrupts) */
    PIE1=0b00000000;            /* disable all peripheral interrupts */
    PIE2=0b00000000;

/* initialize T1CON and T2CON registers */
    T1CON=0b00000000;           /* T1 not used  */
    T2CON=0b00000000;           /* T2 not used  */

/* initialize CCPxCON registers */
    CCP1CON=0b00001100;         /* set CCP1CON for PWM mode */
    CCP2CON=0b00000000;         /* CCP2CON=0 (PWM 2 not used) */

/* initialize SSPCON register */
    SSPCON=0b00000000;          /* serial port - not used */

/* initialize ADCONx registers */
    ADCON0=0b00000000;          /* a-d converter */
    ADCON1=0b00000010;

/* initialize TRISx register (port pins as inputs or outputs) */
    TRISA=0b00001111;
    TRISB=0b00000000;
    TRISC=0b11111011;
    TRISD=0b11111111;
    TRISE=0b11111111;

/* clear watchdog timer (not used) */
    CLRWDT();

/* initialize program bit variables */
   FLAGS=0b00000000;

/* intialize output port pins */
   PORTB=0;

/* enable interrupts...  */

    INTCON.T0IE=1;              /* peripheral interrupt enable */
    INTCON.GIE=1;               /* global interrupt enable */

    init_PWM(0x40);             /* init PWM port */
    init_filter();              /* init filter */
    while(1){
      while(!sample_flag){}     /* wait for sample clock flag to be set */
        sample_flag=0;          /* clear sample clock flag */
        x_n=get_sample(1);      /* read ADC channel 1 into x(n) */
        if(FILTER==1){          /* if filter enabled */
          filter();             /* call filter routine */
        }
        else{                   /* or else write x(n) to y(n) (talkthru) */
          y_n=x_n;
        }
        write_PWM((char)y_n,0); /* write y_n to PWM port 1 */
    }
```

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APPENDIX F: 8-BIT MULTIPLY AND SCALING ROUTINES

```
/************************************************************************
* 8 bit Multiply and Scaling Routines
*
* Written for "Digital Signal Processing with the PIC16C74" Application Note.
*
*
* This module provides a 8 bit signed multiply and scaling routine for the
* PICTONE.C tone generation program. The routines are adapted from "Math
* Routines for the 16C5x" in  Microchip's Embedded Controller Handbook.
*
* All numbers are assumed to be signed 2's compliment format.
*
* D. Mostowfi 11/94
************************************************************************/
char multcnd;                   /* 8 bit multiplicand */
char multplr;                   /* 8 bit multiplier */
char result_h;                  /* result - high byte */
char result_l;                  /* result - low byte */
char sign;                      /* result sign */

#asm

;
; 8x8 signed multiply routine
; called from PICTONE.C module (assembly language routine)
;
 .MACRO mult_core bit
        btfss   multplr,bit
        goto    \no_add
        movf    multcnd,W
        addwf   result_h,F

\no_add:
        rrf     result_h
        rrf     result_l
 .ENDM


_8x8smul:
        movf    multcnd,W       ; get multiplicand
        xorwf   multplr,W       ; and xor with multiplier
        movwf   sign            ; and save sign of result
        btfss   multcnd,7       ; check sign bit of multiplicand
        goto    chk_multplr     ; go and check multipier if positive
        comf    multcnd         ; negate if negative
        incf    multcnd         ;

chk_multplr:
        btfss   multplr,7       ; check sign bit of multiplier
        goto    multiply        ; go to multiply if positive
        comf    multplr         ; negate if negative
        incf    multplr         ;

multiply:
        movf    multcnd,W       ; set up multiply registers
        bcf     STATUS,C        ;
        clrf    result_h        ;
        clrf    result_l        ;
        mult_core 0             ; and do multiply core 8 times
        mult_core 1             ;
```

```
        mult_core 2             ;
        mult_core 3             ;
        mult_core 4             ;
        mult_core 5             ;
        mult_core 6             ;
        mult_core 7             ;

set_sign:
        btfss  sign,7           ; test sign to see if result negative
        retlw  0                ; done if not! (clear W)
        comf   result_l         ; negate result if sign set
        incf   result_l         ;
        btfsc  STATUS,Z         ;
        decf   result_h         ;
        comf   result_h         ;

        retlw  0                ; done (clear W)


;
; Scaling Routine (used after a multiply to scale 16 bit result)
; Operates on result_h and result_l - final result is in result_l
; routine divides by 32 to restore Q7 result of 2*b*y(n-1) in tone
; generation algorithm
;
scale_16:
        btfss  sign,7           ; test if negative (sign set from mult)
        goto   start_shift      ; go to start shift if pos.
        comf   result_l         ; negate first if neg.
        incf   result_l         ;
        btfsc  STATUS,Z         ;
        decf   result_h         ;
        comf   result_h         ;

start_shift:
        bcf    STATUS,C         ; clear status
        rrf    result_h         ; and shift result left 5x (/32)
        rrf    result_l         ;
        rrf    result_h         ;
        rrf    result_l         ;
        rrf    result_h         ;
        rrf    result_l         ;
        rrf    result_h         ;
        rrf    result_l         ;
        rrf    result_h         ;
        rrf    result_l         ;

        btfss  sign,7           ; test if result negative
        goto   scale_done       ; done if not negative
        comf   result_l         ; negate result if negative
        incf   result_l         ;
        btfsc  STATUS,Z         ;
        decf   result_h         ;
        comf   result_h         ;

scale_done:                     ;

        retlw  0                ; done (clear W)

#endasm
```

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APPENDIX G: DOUBLE PRECISION MATH ROUTINES

```
/************************************************************************
* Double Precision Math Routines
*
* This module contains assembly language routines from "Math Routines for the
* 16C5x" from Microchip's Embedded Controller Handbook that have been adapted
* for use with the Bytecraft MPC C Compiler.
*
* Routines are used IIR_FILT.C module written for "Digital Signal Processing
* with the PIC16C74" Application Note.
*
* D. Mostowfi 3/95
************************************************************************/

/*
Start of converted MPASM modules:

;*****************************************************************
;                 Double Precision Addition & Subtraction
;
;*****************************************************************;
;   Addition :  ACCb(16 bits) + ACCa(16 bits) -> ACCb(16 bits)
;      (a) Load the 1st operand in location ACCaLO & ACCaHI ( 16 bits )
;      (b) Load the 2nd operand in location ACCbLO & ACCbHI ( 16 bits )
;      (c) CALL D_add
;      (d) The result is in location ACCbLO & ACCbHI ( 16 bits )
;
;   Performance :
;               Program Memory  :       07
;               Clock Cycles    :       08
;*****************************************************************;
;   Subtraction : ACCb(16 bits) - ACCa(16 bits) -> ACCb(16 bits)
;      (a) Load the 1st operand in location ACCaLO & ACCaHI ( 16 bits )
;      (b) Load the 2nd operand in location ACCbLO & ACCbHI ( 16 bits )
;      (c) CALL D_sub
;      (d) The result is in location ACCbLO & ACCbHI ( 16 bits )
;
;   Performance :
;               Program Memory  :       14
;               Clock Cycles    :       17
;*****************************************************************;
;
*/

char ACCaLO;  //equ     10   changed equ statements to C char variables
char ACCaHI;  //equ     11
char ACCbLO;  //equ     12
char ACCbHI;  //equ     13
;

#asm            /* start of in-line assembly code */

;        include "mpreg.h"     commented out these
;        org     0             two lines (MPASM specific)

;*****************************************************************
;         Double Precision Subtraction ( ACCb - ACCa -> ACCb )
;
D_sub   call    neg_A2           ; At first negate ACCa; Then add
;
;*****************************************************************
```

```
;       Double Precision  Addition ( ACCb + ACCa -> ACCb )
;
D_add   movf    ACCaLO,W
    addwf   ACCbLO              ;add lsb
    btfsc   STATUS,C            ;add in carry
    incf    ACCbHI
    movf    ACCaHI,C
    addwf   ACCbHI              ;add msb
    retlw   0

neg_A2  comf    ACCaLO          ; negate ACCa ( -ACCa -> ACCa )
    incf    ACCaLO
    btfsc   STATUS,Z
    decf    ACCaHI
    comf    ACCaHI
    retlw   0


;*****************************************************************
;                       Double Precision Multiplication
;
;               ( Optimized for Speed : straight Line Code )
;
;*****************************************************************;
;   Multiplication : ACCb(16 bits) * ACCa(16 bits) -> ACCb,ACCc ( 32 bits )
;      (a) Load the 1st operand in location ACCaLO & ACCaHI ( 16 bits )
;      (b) Load the 2nd operand in location ACCbLO & ACCbHI ( 16 bits )
;      (c) CALL D_mpy
;      (d) The 32 bit result is in location ( ACCbHI,ACCbLO,ACCcHI,ACCcLO )
;
;   Performance :
;               Program Memory  :       240
;               Clock Cycles    :       233
;
;       Note : The above timing is the worst case timing, when the
;               register ACCb = FFFF. The speed may be improved if
;               the register ACCb contains a number ( out of the two
;               numbers ) with less number of 1s.
;
;               The performance specs are for Unsigned arithmetic ( i.e,
;               with “SIGNED equ  FALSE “).

;*****************************************************************;
;

#endasm
//char ACCaLO;  equ     10   Commented out - already defined in Dbl_add
//char ACCaHI;  equ     11
//char ACCbLO;  equ     12
//char ACCbHI;  equ     13
char ACCcLO;  //equ     14   changed equ statements to C char variables
char ACCcHI;  //equ     15
char ACCdLO;  //equ     16
char ACCdHI;  //equ     17
char temp;    //equ     18
char sign;    //equ     19

#asm
;
;        include “mpreg.h”       commented out these
;        org     0               two lines (MPASM specific)
;*****************************************************************
SIGNED  equ     1               ; Set This To ‘TRUE’ if the routines
;                               ; for Multiplication & Division needs
;                               ; to be assembled as Signed Integer
;                               ; Routines. If ‘FALSE’ the above two
```

```
;                               ; routines ( D_mpy & D_div ) use
;                               ; unsigned arithmetic.
;*****************************************************************
;       multiplication macro
;
 .MACRO mulMac                  ; changed macro to conform to MPC macro
;        LOCAL   NO_ADD         ; language - declaration is different
;                               ; and macro labels are preceded by “/”

    rrf     ACCdHI              ; rotate d right
    rrf     ACCdLO
    btfss   STATUS,C            ; need to add?
    goto    \NO_ADD             ; no addition necessary
    movf    ACCaLO,W            ; Addition ( ACCb + ACCa -> ACCb )
    addwf   ACCbLO              ; add lsb
    btfsc   STATUS,C            ; add in carry
    incf    ACCbHI
    movf    ACCaHI,W
    addwf   ACCbHI              ;add msb
\NO_ADD rrf     ACCbHI
    rrf     ACCbLO
    rrf     ACCcHI
    rrf     ACCcLO
;
    .ENDM                       ; end of modified macro
;
;*****************************************************************;
;               Double Precision Multiply ( 16x16 -> 32 )
;         ( ACCb*ACCa -> ACCb,ACCc ) : 32 bit output with high word
;  in ACCb ( ACCbHI,ACCbLO ) and low word in ACCc ( ACCcHI,ACCcLO ).
;
D_mpyF                          ;results in ACCb(16 msb’s) and ACCc(16 lsb’s)
;
     .IF   SIGNED
     CALL    S_SIGN
     .ENDIF
;
    call    setup
;
; use the mulMac macro 16 times
;
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
    mulMac
;
    .IF    SIGNED
    btfss   sign,7
    retlw   0
    comf    ACCcLO              ; negate ACCa ( -ACCa -> ACCa )
    incf    ACCcLO
    btfsc   STATUS,Z
    decf    ACCcHI
    comf    ACCcHI
```

```
    btfsc   STATUS,Z
neg_B   comf    ACCbLO          ; negate ACCb
    incf    ACCbLO
    btfsc   STATUS,Z
    decf    ACCbHI
    comf    ACCbHI
    retlw   0
    .ELSE
    retlw   0
    .ENDIF
;
;*****************************************************************
;
setup   movlw   16              ; for 16 shifts
    movwf   temp
    movf    ACCbHI,W            ;move ACCb to ACCd
    movwf   ACCdHI
    movf    ACCbLO,W
    movwf   ACCdLO
    clrf    ACCbHI
    clrf    ACCbLO
    retlw   0
;
;*****************************************************************
;
neg_A   comf    ACCaLO          ; negate ACCa ( -ACCa -> ACCa )
    incf    ACCaLO
    btfsc   STATUS,Z
    decf    ACCaHI
    comf    ACCaHI
    retlw   0
;
;*****************************************************************
;  Assemble this section only if Signed Arithmetic Needed
;
     .IF    SIGNED
;
S_SIGN  movf    ACCaHI,W
    xorwf   ACCbHI,W
    movwf   sign
    btfss   ACCbHI,7            ; if MSB set go & negate ACCb
    goto    chek_A
;
    comf    ACCbLO              ; negate ACCb
    incf    ACCbLO
    btfsc   STATUS,Z
    decf    ACCbHI
    comf    ACCbHI
;
chek_A  btfss   ACCaHI,7        ; if MSB set go & negate ACCa
    retlw   0
    goto    neg_A
;
     .ENDIF

#endasm
```