AN643

Adaptive Differential Pulse Code Modulation using PIC16/17 Microcontrollers

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INTRODUCTION

In the past, adding speech recording and playback capability to a product meant using a digital signal processor or a specialized audio chip. Now, using a simplified Adaptive Differential Pulse Code Modulation (ADPCM) algorithm, these audio capabilities can be added to any PIC16/17 device. This application note will cover the ADPCM compression and decompression algorithms, performance comparison of all PIC16/17 devices, and an application using a PIC16C72 microcontroller.

DEFINITION OF TERMS

step size - value of the step used for quantization of analog signals and inverse quantization of a number of steps.
quantization - the digital form of an analog input signal is represented by a finite number of steps.
adaptive quantization - the step size of a quantizer is dramatically changed with time in order to adapt to a changing input signal.
inverse quantizer - a finite number of steps is converted into a digital representation of an analog signal.

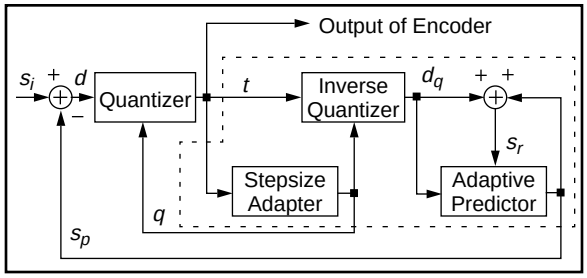
THEORY OF OPERATION

The ADPCM algorithm takes advantage of the high correlation between consecutive speech samples, which enables future sample values to be predicted. Instead of encoding the speech sample, ADPCM encodes the difference between a predicted sample and the speech sample. This method provides more efficient compression with a reduction in the number of bits per sample, yet preserves the overall quality of the speech signal. The implementation of the ADPCM algorithm provided in this application note is based on the Interactive Multimedia Association's (IMA) Recommended Practices for Enhancing Digital Audio Compatibility in Multimedia Systems revision 3.00. The ITU (formerly CCITT) G.721 ADPCM algorithm is well known and has been implemented on many digital signal processors such as the TMS320 family from Texas Instruments and the ADSP-2100 family from Analog Devices. ITU G.721 uses floating point arithmetic and logarithmic functions which are not easily implemented in the 8-bit microcontroller world. The IMA Reference algorithm significantly reduces the mathematical complexity of ITU G.721 by simplifying many of the operations and using table lookups where appropriate.

COMPRESSION

The input, s_i, to the encoder routine must be 16-bit two's complement speech data. The range of allowable values for s_i is 32767 to -32768. Figure 1 shows a block diagram for ADPCM compression and Appendix A has a listing of the function **ADPCMEncoder()**. The predicted sample, s_p, and the quantizer step size index are saved in a structure for the next iteration of the encoder. Initially, the quantizer step size index and the predicted sample (s_p) are set to zero. The encoder function takes a 16-bit two's complement speech sample and returns an 8-bit number containing the 4-bit sign-magnitude ADPCM code.

FIGURE 1: ADPCM ENCODER BLOCK DIAGRAM



The predicted sample, s_p, is subtracted from the linear input sample, s_i, to produce a difference, *d*. Adaptive quantization is performed on the difference, resulting in the 4-bit ADPCM value, *t*. The encoder and decoder both update their internal variables based on this ADPCM value. A full decoder is actually embedded within the encoder. This ensures that the encoder and decoder are synchronized without the need to send any additional data. The embedded decoder is shown within the dotted lines of Figure 1. The embedded decoder uses the ADPCM value to update the inverse quantizer, which produces a dequantized version, d_q, of the difference, *d*. The implementation of the ADPCM algorithm presented in this application note uses a

fixed predictor instead of an adaptive predictor which reduces the amount of data memory and instruction cycles required.

The adaptive predictor of ITU G.721 adjusts according to the value of each input sample using a weighted average of the last six dequantized difference values and the last two predicted values. So at this point, the dequantized difference, d_q, is added to the predicted sample, s_p, to produce a new predicted sample, s_r. Finally the new predicted sample, s_r, is saved into s_p.

The following (Table 1) is a step-by-step description of the **ADPCMEncoder()** function from Appendix A.

TABLE 1: ADPCMEncoder() STEP-BY-STEP FUNCTIONS

1. ADPCMEncoder takes a 16-bit signed number (speech sample, 32767 to -32768) and returns an 8-bit number containing the 4-bit ADPCM code (0-15)	`char ADPCMEncoder(long signed sample)`
2. Restore the previous values of predicted sample (s_p) and the quantizer step size index	`predsample = state.prevsample;` `index = state.previndex;`
3. Find the quantizer step size (q) from a table lookup using the quantizer step size index	`step = StepSizeTable[index];`
4. Compute the difference (d) between the actual sample (s_i) and the predicted sample (s_p)	`diff = sample - predsample;`

5. Set the sign bit of the ADPCM code (t) if necessary and find the absolute value of difference (d)	`if(diff >= 0)`
	`  code = 0;`
	`else`
	`{`
	`  code = 8;`
	`  diff = - diff;`
	`}`

6. Save quantizer step size (q) in a temporary variable	`tempstep = step;`

7. Quantize the difference (d) into the ADPCM code (t) using the quantizer step size (q)	`if(diff >= tempstep)`
	`{`
	`  code |= 4;`
	`  diff -= tempstep;`
	`}`
	`tempstep >>= 1;`
	`if(diff >= tempstep)`
	`{`
	`  code |= 2;`
	`  diff -= tempstep;`
	`}`
	`tempstep >>= 1;`
	`if(diff >= tempstep)`
	`  code |= 1;`

8. Inverse quantize the ADPCM code (t) into a predicted difference (d_q) using the quantizer step size (q)	`diffq = step >> 3;`
	`if(code & 4)`
	`  diffq += step;`
	`if(code & 2)`
	`  diffq += step >> 1;`
	`if(code & 1)`
	`  diffq += step >> 2;`

TABLE 1: ADPCMEncoder() STEP-BY-STEP FUNCTIONS (CONTINUED)

9. Fixed predictor computes new predicted sample (s_r) by adding the old predicted sample (s_p) to the predicted difference (d_q)	`if(code & 8)`
	`  predsample -= diffq;`
	`else`
	`  predsample += diffq;`

10. Check for overflow of the new predicted sample (s_r). s_r, which is a signed 16-bit sample, must be in the range of 32767 to -32768	`if(predsample > 32767)`
	`  predsample = 32767;`
	`else if(predsample < -32768)`
	`  predsample = -32768;`

11. Find the new quantizer step size index (q) by adding the previous index and a table lookup using the ADPCM code (t)	`index += IndexTable[code];`

12. Check for overflow of the new quantizer step size index	`if(index < 0)`
	`  index = 0;`
	`if(index > 88)`
	`  index = 88;`

13. Save the new predicted sample (s_r) and quantizer step size index for next iteration	`state.prevsample = predsample;` `state.previndex = index;`
14. Return the ADPCM code (t)	`return (code & 0x0f);`

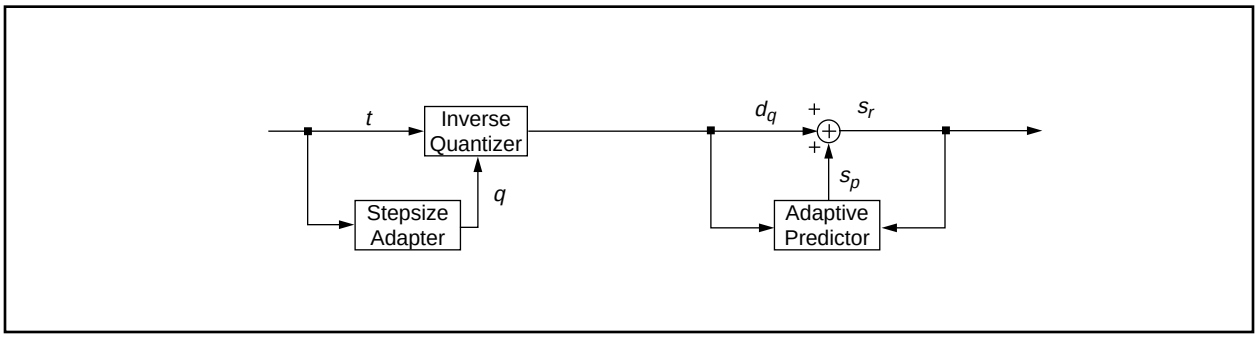
This function requires five 16-bit variables and two 8-bit variables. Some optimizations can be made to the code in Appendix A such as combining steps 7 and 8. Appendix C gives the output listing of the MPC compiler for a PIC16CXX device using the optimized encoder algorithm and the decoder algorithm from the next section.

DECOMPRESSION

The input into the decoder, t, must be an 8-bit number containing the 4-bit ADPCM data in sign-magnitude format. The range of allowable values for t is 0 to 15, where 7 = 0x07 and -7 = 0x0F. Figure 2 shows a block diagram for ADPCM decompression and Appendix B has a listing of the function **ADPCMDecoder()**. The predicted sample, s_p, and the quantizer step size index are saved in a structure for the next iteration of the decoder. Initially, the quantizer step size index and the predicted sample (s_p) are set to zero. This function takes a 4-bit sign-magnitude ADPCM code and returns the 16-bit two's complement speech sample.

This decoder is the same as the one used in the encoder routine. It uses the ADPCM value to update the inverse quantizer, which produces a difference, d_q. The difference, d_q, is added to the predicted sample, s_p, to produce the output sample, s_r. The output sample, s_r, is then saved into the predicted sample, s_p, for the next iteration of the decoder.

FIGURE 2: ADPCM DECODER BLOCK DIAGRAM



The following (Table 2) is a step-by-step description of the **ADPCMDecoder()** function from Appendix B.

TABLE 2: ADPCMDecoder() STEP-BY-STEP FUNCTIONS

1. ADPCMDecoder takes an 8-bit number containing the 4-bit ADPCM code (0-15) and returns a 16-bit signed number (speech sample, 32767 to -32768)	`signed long ADPCMDecoder( char code )`
2. Restore the previous values of predicted sample (s_p) and quantizer step size index	`predsample = state.prevsample;` `index = state.previndex;`
3. Find the quantizer step size (q) from a table lookup using the quantizer step size index	`step = StepSizeTable[index];`

4. Inverse quantize the ADPCM code (t) into a predicted difference (d_q) using the quantizer step size (q)	`diffq = step >> 3;`
	`if(code & 4)`
	`  diffq += step;`
	`if(code & 2)`
	`  diffq += step >> 1;`
	`if(code & 1)`
	`  diffq += step >> 2;`

5. Fixed predictor computes new predicted sample (s_r) by adding the old predicted sample (s_p) to the predicted difference (d_q)	`if(code & 8)`
	`  predsample -= diffq;`
	`else`
	`  predsample += diffq;`

6. Check for overflow of the new predicted sample (s_r). s_r, which is a signed 16-bit sample, must be in the range of 32767 to -32768	`if(predsample > 32767)`
	`  predsample = 32767;`
	`else if(predsample < -32768)`
	`  predsample = -32768;`

7. Find the new quantizer step size (q) by adding the previous index and a table lookup using the ADPCM code (t)	`index += IndexTable[code];`

8. Check for overflow of the new quantizer step size index	`if(index < 0)`
	`  index = 0;`
	`if(index > 88)`
	`  index = 88;`

9. Save the new predicted sample (s_r) and quantizer step size index for next iteration	`state.prevsample = predsample;` `state.previndex = index;`
10. Return the new sample (s_r)	`return ( predsample );`

This function requires three 16-bit variables and one 8-bit variable. Appendix C gives the listing output of the MP-C compiler of the decoder algorithm and the optimized encoder algorithm.

IMA ADPCM REFERENCE ALGORITHM

The IMA, specifically the Digital Audio Technical Working Group, is a trade association with representatives from companies such as Compaq®, Apple Computer®, Crystal Semiconductor®, DEC®, Hewlett-Packard®, Intel®, Microsoft®, Sony®, and Texas Instruments® to name a few. This group is working towards a standard that defines the exchange of high quality audio data between computing platforms. The algorithm from Intel/DVI® (Digital Video Interactive) has been selected as the standard due to it's audio dynamic range and low data rate. The recommended digital audio exchange formats are given in Table 3.

The algorithms that are implemented in this application note were derived from the IMA ADPCM Reference algorithm. The data format is 8.0 kHz, mono, 4-bit ADPCM. Essentially, the compression and decompression use an adaptive quantization with fixed prediction. The adaptive quantization are based on a table lookup first developed by Intel/DVI for the IMA. Appendix D gives the information about IMA and the supporting documentation for IMA ADPCM.

PERFORMANCE

Table 4 shows the performance comparison of the ADPCM compression and decompression routines for the PIC16C5X, PIC16CXX, and PIC17CXX family of devices assuming 80 additional instruction cycles for overhead. Any device without a PWM module will have to increase the operating frequency to generate a software PWM. Table 4 also provides the minimum external operating frequency required to run the routines. The C code from Appendix C was used. The input/output data rate of the speech samples is 8.0 kHz. The minimum operating frequency is calculated as follows:

8000 x # of Total Instruction Cycles x 4.

For example, the ADPCM encoding using the PIC16CXX family would require an 8000 x (225 + 80) x 4 = 9.760 MHz external crystal.

TABLE 3: DIGITAL AUDIO EXCHANGE FORMATS

Sampling Rate	Mono/Stereo	Data Format	Notes
8.0 kHz	mono	8-bit μ-Law PCM	CCITT G.711 Standard
	mono	8-bit A-Law PCM	CCITT G.711 Standard
	mono	4-bit ADPCM	DVI Algorithm
11.025 kHz	mono/stereo	8-bit Linear PCM	Macintosh® & MP-C Standard
	mono/stereo	4-bit ADPCM	DVI Algorithm
22.05 kHz	mono/stereo	8-bit Linear PCM	Macintosh & MPC Standard
	mono/stereo	4-bit ADPCM	DVI Algorithm
44.10 kHz	mono/stereo	16-bit Linear PCM	CD-DA Standard
	mono/stereo	4-bit ADPCM	DVI Algorithm

TABLE 4: PERFORMANCE COMPARISON TABLE

Device	Encode/Decode	# of Instruction Cycles	Minimum External Operating Frequency
PIC16C5X	Encode	273	11.296 MHz
	Decode	208	9.216 MHz
PIC16CXX	Encode	225	9.760 MHz
	Decode	168	7.936 MHz
PIC17CXX	Encode	199	8.928 MHz
	Decode	161	7.712 MHz

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Table 5 illustrates the amount of program and data memory that the ADPCM algorithms consume for the various PIC16/17 devices. The program memory numbers may change slightly depending on the specific device being used. The table memory column shows how much program memory is used to store the two lookup tables used by the ADPCM algorithm.

TABLE 5: DEVICE MEMORY CONSUMED

Device		Program Memory (words)	Data Memory (bytes)	Table Lookup (words)
PIC16C5X	Encode	273	13	196
	Decode	205	10	
PIC16CXX	Encode	220	13	196
	Decode	162	10	
PIC17CXX	Encode	203	13	97
	Decode	164	10	

APPLICATION

The hardware for this application note implements only the decompression algorithm, but the compression algorithm is also included in the firmware. A block diagram is shown in Figure 3 and the schematic is provided in Appendix E. The complete source code listing, including a block diagram, is given in Appendix F.

The board uses a PIC16C72, rated for 20 MHz operation, to control the speech decoding and output. The A/D (Analog-to-Digital) converter of the PIC16C72 is not used in this design. Two 27C512A EPROMs from Microchip Technology are used to store up to 32.768 seconds of ADPCM data. Each EPROM holds 65536 bytes. Each byte in the EPROM contains two ADPCM values. One second of speech requires 8000 ADPCM codes (8 kHz sample rate). Therefore, each EPROM holds (65536 x 2) / 8000 = 16.384 seconds of speech. A 16-bit up counter is used to clock data out of the EPROMs. This method uses only two I/O lines to control the counter and eleven I/O lines to read the EPROM data.

Speech regeneration is accomplished by using the Capture/Compare/PWM (CCP) module of the PIC16C72. The PWM module is configured for a period of 32 kHz. This allows each sample of the speech signal to be output for four PWM periods. The period is calculated by using the following equation from Section 10.3 of the PIC16C7X Data Sheet (DS30390). From the following calculation, PR2 is set to a value of 155.

PWM period = [PR2 + 1] * 4 * Tosc * (TMR2 prescale value)

1 / 32 kHz = [PR2 + 1] * 4 * (1 / 20 MHz) * 1
31.25 µs = [PR2 + 1] * 4 * 50 ns * 1
156.25 = PR2 + 1
155.25 = PR2

The CCP module has the capability of up to 10-bit resolution for the PWM duty cycle. The maximum resolution of the duty cycle that can be achieved for a 32 kHz period can be calculated using the following equation from Section 10.3 of the PIC16C7X Data Sheet.

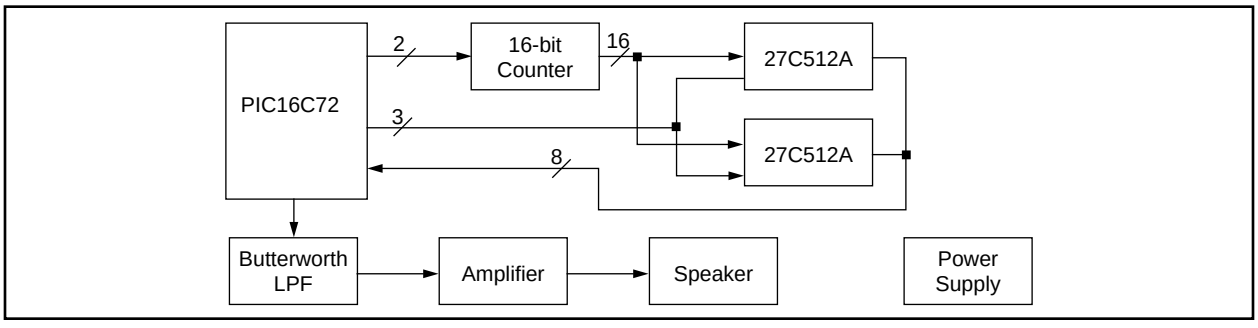
PWM duty cycle = DC<10:0> * Tosc * (TMR2 prescale value)

where DC<10:0> = 2^x where x = bits of resolution

1 / 32 kHz = 2^x * (1 / 20 MHz) * 1
31.25 µs = 2^x * 50 ns * 1
625 = 2^x
log(625) = log(2^x)
log(625) = x * log(2)
9.3 = x

A PWM duty cycle with up to 9-bits of resolution may be used with a period of 32 kHz. The upper 9-bits of each speech sample are used to set the PWM duty cycle (CCPR1L = sample<15:9>, CCP1CON<5:4> = sample<8:7>). This PWM speech signal is passed through a 4th-order Butterworth low pass filter with a corner frequency of 4 kHz. The low pass filter converts the PWM into an analog voltage level. Finally, the analog voltage level is amplified by a National Semiconductor LM386N-3 before entering the speaker. Every 125 μs (8 kHz) one ADPCM code must be converted into a speech sample for output to the PWM. This frame of 125 μs relates to 625 instruction cycles for an external crystal frequency of 20 MHz. The ADPCM decode routine, EPROM reads and writes to the PWM must be accomplished within 625 instruction cycles.

FIGURE 3: APPLICATION HARDWARE BLOCK DIAGRAM



To implement audio recording, the following changes to the application hardware would have to be made:

- Replace the EPROMs with RAM and
- Add a microphone with input filtering and
- Use the on-board A/D converter of the PIC16C7X devices for 8-bit recording or
- Use an external 12- to 16-bit A/D converter for better quality recording

The following is the sequence of events to record and store speech data. A timer would be used to set the sample rate. When the timer overflowed an A/D conversion would be started.

1. Start an A/D conversion when the timer overflows.
2. Read the sample and call the routine ADPCMEncoder().
3. Store the result in the upper 4-bits of a temporary variable.
4. Start an A/D conversion when the timer overflows.
5. Read the sample and call the routine ADPCMEncoder().
6. Store the result in the lower 4-bits of the temporary variable.
7. Write the temporary variable to the SRAM.
8. GOTO 1.

Typical conversion times for the PIC16C7X devices are 40 μs. Assuming that the external crystal frequency is 20 MHz and the sample rate is 8 kHz, the encoder routine takes approximately 45 μs (225 instruction cycles x 700 μs) to complete. This leaves approximately 40 μs or 200 instruction cycles to read the A/D converter, write to the SRAM and complete any other tasks. An external A/D converter that is faster could be used to increase the amount of time for processing other tasks.

COMPUTER PROGRAM

The IMA Reference ADPCM algorithm was written for a PC using Borland C++ version 3.1. This program uses the same routines that are used in Appendix C. A raw sound file is recorded on the PC using a 16-bit sound card. Speech is recorded at 8.0 kHz in 16-bit mono mode. This raw speech file is processed by the encoder part of the program which creates a file with the ADPCM values. This file can now be burned into EPROMs for use with the application hardware. The decoder part of the program can take this file and create a new 16-bit raw speech file. The 16-bit sound card on the PC can play this file to ensure that the ADPCM compression/decompression routines are working properly. The PC program is given in Appendix F. It is composed of three files: pcspeech.c, pcadpcm.c, and pcadpcm.h.

CONCLUSION

The final results of the application hardware using the PIC16C72 are:

- Compression and Decompression
  - 906 words of Program Memory used
  - 44% of total Program Memory
  - 24 bytes of Data Memory used
  - 19% of total Data Memory
- Decompression only
  - 686 words of Program Memory used
  - 33% of total Program Memory
  - 22 bytes of Data Memory used
  - 17% of total Data Memory
- ~250 Instruction cycles to decode one ADPCM code
  - 40% of frame (250/625)
- Hardware used
  - CCP1 configured for PWM (9-bit duty cycle, 32 kHz period speech output)
  - Timer2 to set the 8.0 kHz output sample rate
  - 13 I/O lines to read data from the EPROMs

References:

1. *Recommended Practices for Enhancing Digital Audio Compatibility in Multimedia Systems*, Revision 3.00, Interactive Multimedia Association, October 21, 1992.
2. *Digital Audio Special Edition Proceedings*, Volume 2, Issue 2, Interactive Multimedia Association, May 1992.
3. Panos E. Papamichalis Ph.D., *Practical Approaches to Speech Coding*, Prentice-Hall Inc., Englewood Cliffs, N.J, 1987.
4. Adaptive Differential Pulse-Code Modulation, *Digital Signal Processing Applications using the ADSP-2100 Family*, Volume 1, Analog Devices, Prentice-Hall, Englewood Cliffs, N.J., 1992.
5. 32-kbits/s ADPCM with the TMS32010, *Digital Signal Processing Applications with the TMS320 Family*, SPRA012, Jay Reimer, Mike McMahan, and Masud Arjmand, Texas Instruments, 1986.

APPENDIX A: ADPCMEncoder() FUNCTION

```
/* Table of index changes */
const int IndexTable[16] = {
    0xff, 0xff, 0xff, 0xff, 2, 4, 6, 8,
    0xff, 0xff, 0xff, 0xff, 2, 4, 6, 8
};

/* Quantizer step size lookup table */
const long StepSizeTable[89] = {
    7, 8, 9, 10, 11, 12, 13, 14, 16, 17,
    19, 21, 23, 25, 28, 31, 34, 37, 41, 45,
    50, 55, 60, 66, 73, 80, 88, 97, 107, 118,
    130, 143, 157, 173, 190, 209, 230, 253, 279, 307,
    337, 371, 408, 449, 494, 544, 598, 658, 724, 796,
    876, 963, 1060, 1166, 1282, 1411, 1552, 1707, 1878, 2066,
    2272, 2499, 2749, 3024, 3327, 3660, 4026, 4428, 4871, 5358,
    5894, 6484, 7132, 7845, 8630, 9493, 10442, 11487, 12635, 13899,
    15289, 16818, 18500, 20350, 22385, 24623, 27086, 29794, 32767
};

signed long diff;               /* Difference between sample and predicted sample */
long step;                      /* Quantizer step size */
signed long predsample;         /* Output of ADPCM predictor */
signed long diffq;              /* Dequantized predicted difference */
int index;                      /* Index into step size table */

/*****************************************************************************
*       ADPCMEncoder - ADPCM encoder routine                                 *
*****************************************************************************
*       Input Variables:                                                     *
*               signed long sample - 16-bit signed speech sample             *
*       Return Variable:                                                     *
*               char - 8-bit number containing the 4-bit ADPCM code          *
*****************************************************************************/
char ADPCMEncoder( signed long sample )
{
    int code;               /* ADPCM output value */
    int tempstep;           /* Temporary step size */

    /* Restore previous values of predicted sample and quantizer step
       size index
    */
    predsample = state.prevsample;
    index = state.previndex;
    step = StepSizeTable[index];


    /* Compute the difference between the actual sample (sample) and the
       the predicted sample (predsample)
    */
    diff = sample - predsample;
    if(diff >= 0)
        code = 0;
    else
    {
        code = 8;
        diff = -diff;
    }

    /* Quantize the difference into the 4-bit ADPCM code using the
       the quantizer step size
    */
    tempstep = step;
    if( diff >= tempstep )
    {
        code |= 4;
```

```
        diff -= tempstep;
    }
    tempstep >>= 1;
    if( diff >= tempstep )
    {
        code |= 2;
        diff -= tempstep;
    }
    tempstep >>= 1;
    if( diff >= tempstep )
        code |= 1;

    /* Inverse quantize the ADPCM code into a predicted difference
       using the quantizer step size
    */
    diffq = step >> 3;
    if( code & 4 )
        diffq += step;
    if( code & 2 )
        diffq += step >> 1;
    if( code & 1 )
        diffq += step >> 2;

    /* Fixed predictor computes new predicted sample by adding the
       old predicted sample to predicted difference
    */
    if( code & 8 )
        predsample -= diffq;
    else
        predsample += diffq;

    /* Check for overflow of the new predicted sample
    */
    if( predsample > 32767 )
        predsample = 32767;
    else if( predsample < -32768 )
        predsample = -32768;

    /* Find new quantizer stepsize index by adding the old index
       to a table lookup using the ADPCM code
    */
    index += IndexTable[code];

    /* Check for overflow of the new quantizer step size index
    */
    if( index < 0 )
        index = 0;
    if( index > 88 )
        index = 88;

    /* Save the predicted sample and quantizer step size index for
       next iteration
    */
    state.prevsample = predsample;
    state.previndex = index;

    /* Return the new ADPCM code */
    return ( code & 0x0f );
}
```

APPENDIX B: ADPCMDecoder() FUNCTION

```
/* Table of index changes */
const int IndexTable[16] = {
    0xff, 0xff, 0xff, 0xff, 2, 4, 6, 8,
    0xff, 0xff, 0xff, 0xff, 2, 4, 6, 8
};

/* Quantizer step size lookup table */
const long StepSizeTable[89] = {
    7, 8, 9, 10, 11, 12, 13, 14, 16, 17,
    19, 21, 23, 25, 28, 31, 34, 37, 41, 45,
    50, 55, 60, 66, 73, 80, 88, 97, 107, 118,
    130, 143, 157, 173, 190, 209, 230, 253, 279, 307,
    337, 371, 408, 449, 494, 544, 598, 658, 724, 796,
    876, 963, 1060, 1166, 1282, 1411, 1552, 1707, 1878, 2066,
    2272, 2499, 2749, 3024, 3327, 3660, 4026, 4428, 4871, 5358,
    5894, 6484, 7132, 7845, 8630, 9493, 10442, 11487, 12635, 13899,
    15289, 16818, 18500, 20350, 22385, 24623, 27086, 29794, 32767
};

long step;                      /* Quantizer step size */
signed long predsample;         /* Output of ADPCM predictor */
signed long diffq;              /* Dequantized predicted difference */
int index;                      /* Index into step size table */

/*****************************************************************************
*       ADPCMDecoder - ADPCM decoder routine                                 *
*****************************************************************************
*       Input Variables:                                                     *
*               char code - 8-bit number containing the 4-bit ADPCM code     *
*       Return Variable:                                                     *
*               signed long - 16-bit signed speech sample                    *
*****************************************************************************/
signed long ADPCMDecoder(char code )
{

    /* Restore previous values of predicted sample and quantizer step
       size index
    */
    predsample = state.prevsample;
    index = state.previndex;

    /* Find quantizer step size from lookup table using index
    */
    step = StepSizeTable[index];

    /* Inverse quantize the ADPCM code into a difference using the
       quantizer step size
    */
    diffq = step >> 3;
    if( code & 4 )
        diffq += step;
    if( code & 2 )
        diffq += step >> 1;
    if( code & 1 )
        diffq += step >> 2;

    /* Add the difference to the predicted sample
    */
    if( code & 8 )
        predsample -= diffq;
    else
        predsample += diffq;

    /* Check for overflow of the new predicted sample
```

```
    */
    if( predsample > 32767 )
        predsample = 32767;
    else if( predsample < -32768 )
        predsample = -32768;

    /* Find new quantizer step size by adding the old index and a
       table lookup using the ADPCM code
    */
    index += IndexTable[code];

    /* Check for overflow of the new quantizer step size index
    */
    if( index < 0 )
        index = 0;
    if( index > 88 )
        index = 88;


    /* Save predicted sample and quantizer step size index for next
       iteration
    */
    state.prevsample = predsample;
    state.previndex = index;

    /* Return the new speech sample */
    return( predsample );

}
```

APPENDIX C: OPTIMIZED ADPCMEncoder(), ADPCMDecoder(), AND TABLE LISTING

```
/************************************************************************
*    Filename:   ADPCM.C                                                *
************************************************************************
*    Author:     Rodger Richey                                          *
*    Company:    Microchip Technology Incorporated                      *
*    Revision:   0                                                      *
*    Date:       1-9-96                                                 *
*    Compiled using Bytecraft Ltd. MPC version BC.193                   *
************************************************************************
*    This file contains the ADPCM encode and decode routines. This      *
*    routines were obtained from the Interactive Multimedia Association's *
*    Reference ADPCM algorithm. This algorithm was first implemented by *
*    Intel/DVI.                                                         *
************************************************************************/
                                /* Table of index changes */
                                const int IndexTable[16] = {0xff,0xff,0xff,0xff,2,4,6,8,
                                    0xff, 0xff, 0xff, 0xff, 2, 4, 6, 8
0005 0782    ADDWF  PCL         };
0006 34FF    RETLW  FFh
0007 34FF    RETLW  FFh
0008 34FF    RETLW  FFh
0009 34FF    RETLW  FFh
000A 3402    RETLW  02h
000B 3404    RETLW  04h
000C 3406    RETLW  06h
000D 3408    RETLW  08h
000E 34FF    RETLW  FFh
000F 34FF    RETLW  FFh
0010 34FF    RETLW  FFh
0011 34FF    RETLW  FFh
0012 3402    RETLW  02h
0013 3404    RETLW  04h
0014 3406    RETLW  06h
0015 3408    RETLW  08h
                                /* Quantizer step size lookup table */
                                const long StepSizeTable[89] = {
                                    7,8,9,10,11,12,13,14,16,17,19,21,23,25,28,31,34,37,
                                    41,45,50,55,60,66,73,80,88,97,107,118,130,143,157,
                                    173,190,209,230,253,279,307,337,371,408,449,494,544,
                                    598,658,724,796,876,963,1060,1166,1282,1411,1552,
                                    1707,1878,2066,2272,2499,2749,3024,3327,3660,4026,
                                    4428,4871,5358,5894,6484,7132,7845,8630,9493,10442,
                                    11487,12635,13899,15289,16818,18500,20350,22385,
                                    24623, 27086, 29794, 32767
0016 0782    ADDWF  PCL         };
0017 3407    RETLW  07h
0018 3400    RETLW  00h
0019 3408    RETLW  08h
001A 3400    RETLW  00h
001B 3409    RETLW  09h
001C 3400    RETLW  00h
001D 340A    RETLW  0Ah
001E 3400    RETLW  00h
001F 340B    RETLW  0Bh
0020 3400    RETLW  00h
0021 340C    RETLW  0Ch
0022 3400    RETLW  00h
0023 340D    RETLW  0Dh
0024 3400    RETLW  00h
0025 340E    RETLW  0Eh
0026 3400    RETLW  00h
0027 3410    RETLW  10h
0028 3400    RETLW  00h
0029 3411    RETLW  11h
002A 3400    RETLW  00h
```

```
002B 3413    RETLW  13h
002C 3400    RETLW  00h
002D 3415    RETLW  15h
002E 3400    RETLW  00h
002F 3417    RETLW  17h
0030 3400    RETLW  00h
0031 3419    RETLW  19h
0032 3400    RETLW  00h
0033 341C    RETLW  1Ch
0034 3400    RETLW  00h
0035 341F    RETLW  1Fh
0036 3400    RETLW  00h
0037 3422    RETLW  22h
0038 3400    RETLW  00h
0039 3425    RETLW  25h
003A 3400    RETLW  00h
003B 3429    RETLW  29h
003C 3400    RETLW  00h
003D 342D    RETLW  2Dh
003E 3400    RETLW  00h
003F 3432    RETLW  32h
0040 3400    RETLW  00h
0041 3437    RETLW  37h
0042 3400    RETLW  00h
0043 343C    RETLW  3Ch
0044 3400    RETLW  00h
0045 3442    RETLW  42h
0046 3400    RETLW  00h
0047 3449    RETLW  49h
0048 3400    RETLW  00h
0049 3450    RETLW  50h
004A 3400    RETLW  00h
004B 3458    RETLW  58h
004C 3400    RETLW  00h
004D 3461    RETLW  61h
004E 3400    RETLW  00h
004F 346B    RETLW  6Bh
0050 3400    RETLW  00h
0051 3476    RETLW  76h
0052 3400    RETLW  00h
0053 3482    RETLW  82h
0054 3400    RETLW  00h
0055 348F    RETLW  8Fh
0056 3400    RETLW  00h
0057 349D    RETLW  9Dh
0058 3400    RETLW  00h
0059 34AD    RETLW  ADh
005A 3400    RETLW  00h
005B 34BE    RETLW  BEh
005C 3400    RETLW  00h
005D 34D1    RETLW  D1h
005E 3400    RETLW  00h
005F 34E6    RETLW  E6h
0060 3400    RETLW  00h
0061 34FD    RETLW  FDh
0062 3400    RETLW  00h
0063 3417    RETLW  17h
0064 3401    RETLW  01h
0065 3433    RETLW  33h
0066 3401    RETLW  01h
0067 3451    RETLW  51h
0068 3401    RETLW  01h
0069 3473    RETLW  73h
006A 3401    RETLW  01h
006B 3498    RETLW  98h
006C 3401    RETLW  01h
006D 34C1    RETLW  C1h
006E 3401    RETLW  01h
006F 34EE    RETLW  EEh
```

```
0070 3401    RETLW  01h
0071 3420    RETLW  20h
0072 3402    RETLW  02h
0073 3456    RETLW  56h
0074 3402    RETLW  02h
0075 3492    RETLW  92h
0076 3402    RETLW  02h
0077 34D4    RETLW  D4h
0078 3402    RETLW  02h
0079 341C    RETLW  1Ch
007A 3403    RETLW  03h
007B 346C    RETLW  6Ch
007C 3403    RETLW  03h
007D 34C3    RETLW  C3h
007E 3403    RETLW  03h
007F 3424    RETLW  24h
0080 3404    RETLW  04h
0081 348E    RETLW  8Eh
0082 3404    RETLW  04h
0083 3402    RETLW  02h
0084 3405    RETLW  05h
0085 3483    RETLW  83h
0086 3405    RETLW  05h
0087 3410    RETLW  10h
0088 3406    RETLW  06h
0089 34AB    RETLW  ABh
008A 3406    RETLW  06h
008B 3456    RETLW  56h
008C 3407    RETLW  07h
008D 3412    RETLW  12h
008E 3408    RETLW  08h
008F 34E0    RETLW  E0h
0090 3408    RETLW  08h
0091 34C3    RETLW  C3h
0092 3409    RETLW  09h
0093 34BD    RETLW  BDh
0094 340A    RETLW  0Ah
0095 34D0    RETLW  D0h
0096 340B    RETLW  0Bh
0097 34FF    RETLW  FFh
0098 340C    RETLW  0Ch
0099 344C    RETLW  4Ch
009A 340E    RETLW  0Eh
009B 34BA    RETLW  BAh
009C 340F    RETLW  0Fh
009D 344C    RETLW  4Ch
009E 3411    RETLW  11h
009F 3407    RETLW  07h
00A0 3413    RETLW  13h
00A1 34EE    RETLW  EEh
00A2 3414    RETLW  14h
00A3 3406    RETLW  06h
00A4 3417    RETLW  17h
00A5 3454    RETLW  54h
00A6 3419    RETLW  19h
00A7 34DC    RETLW  DCh
00A8 341B    RETLW  1Bh
00A9 34A5    RETLW  A5h
00AA 341E    RETLW  1Eh
00AB 34B6    RETLW  B6h
00AC 3421    RETLW  21h
00AD 3415    RETLW  15h
00AE 3425    RETLW  25h
00AF 34CA    RETLW  CAh
00B0 3428    RETLW  28h
00B1 34DF    RETLW  DFh
00B2 342C    RETLW  2Ch
00B3 345B    RETLW  5Bh
00B4 3431    RETLW  31h
```

```
00B5 344B    RETLW  4Bh
00B6 3436    RETLW  36h
00B7 34B9    RETLW  B9h
00B8 343B    RETLW  3Bh
00B9 34B2    RETLW  B2h
00BA 3441    RETLW  41h
00BB 3444    RETLW  44h
00BC 3448    RETLW  48h
00BD 347E    RETLW  7Eh
00BE 344F    RETLW  4Fh
00BF 3471    RETLW  71h
00C0 3457    RETLW  57h
00C1 342F    RETLW  2Fh
00C2 3460    RETLW  60h
00C3 34CE    RETLW  CEh
00C4 3469    RETLW  69h
00C5 3462    RETLW  62h
00C6 3474    RETLW  74h
00C7 34FF    RETLW  FFh
00C8 347F    RETLW  7Fh

002E                             signed long diff; /* Difference between sample and
                                                      predicted sample */
0030                             long step;                    /* Quantizer step size */
0032                             signed long predsample;  /* Output of ADPCM predictor */
0034                             signed long diffq;/* Dequantized predicted difference */
0036                             int index;             /* Index into step size table */

/************************************************************************
*    ADPCMEncoder - ADPCM encoder routine                               *
************************************************************************
*    Input Variables:                                                  *
*       signed long sample - 16-bit signed speech sample               *
*    Return Variable:                                                  *
*       char - 8-bit number containing the 4-bit ADPCM code            *
************************************************************************/
                                 char ADPCMEncoder( signed long sample )
0037                             {
00C9 1283    BCF    STATUS,RP0
00CA 00B7    MOVWF  37
00CB 0804    MOVF   FSR,W
00CC 00B8    MOVWF  38
0039                                 int code;              /* ADPCM output value */

                                     /* Restore previous values of predicted sample and
                                        quantizer step size index */
00CD 082B    MOVF   2B,W             predsample = state.prevsample;
00CE 00B2    MOVWF  32
00CF 082C    MOVF   2C,W
00D0 00B3    MOVWF  33
00D1 082D    MOVF   2D,W             index = state.previndex;
00D2 00B6    MOVWF  36
00D3 0836    MOVF   36,W             step = StepSizeTable[index];
00D4 0084    MOVWF  FSR
00D5 1003    BCF    STATUS,C
00D6 0D84    RLF    FSR
00D7 3000    MOVLW  00h
00D8 008A    MOVWF  PCLATH
00D9 0804    MOVF   FSR,W
00DA 2016    CALL   0016h
00DB 1283    BCF    STATUS,RP0
00DC 00B0    MOVWF  30
00DD 0A84    INCF   FSR
00DE 3000    MOVLW  00h
00DF 008A    MOVWF  PCLATH
00E0 0804    MOVF   FSR,W
00E1 2016    CALL   0016h
00E2 1283    BCF    STATUS,RP0
00E3 00B1    MOVWF  31
```

```
                                    /* Compute the difference between the actual sample
                                      (sample) and the predicted sample (predsample) */
00E4 0832    MOVF   32,W            diff = sample - predsample;
00E5 0237    SUBWF  37,W
00E6 00AE    MOVWF  2E
00E7 0833    MOVF   33,W
00E8 1C03    BTFSS  STATUS,C
00E9 3E01    ADDLW  01h
00EA 0238    SUBWF  38,W
00EB 00AF    MOVWF  2F
00EC 1BAF    BTFSC  2F,7            if(diff >= 0)
00ED 28F0    GOTO   00F0h
00EE 01B9    CLRF   39                 code = 0;
00EF 28F9    GOTO   00F9h           else
                                    {
00F0 3008    MOVLW  08h                code = 8;
00F1 00B9    MOVWF  39
00F2 09AF    COMF   2F                 diff = -diff;
00F3 09AE    COMF   2E
00F4 0AAE    INCF   2E
00F5 1D03    BTFSS  STATUS,Z
00F6 28F9    GOTO   00F9h
00F7 1283    BCF    STATUS,RP0
00F8 0AAF    INCF   2F
                                    }
                                    /* Quantize the difference into the 4-bit ADPCM code
                                       using the quantizer step size. Inverse quantize
                                       the ADPCM code into a predicted difference using
                                       the quantizer step size */
00F9 1283    BCF    STATUS,RP0      diffq = step >> 3;
00FA 0830    MOVF   30,W
00FB 00B4    MOVWF  34
00FC 0831    MOVF   31,W
00FD 00B5    MOVWF  35
00FE 0D35    RLF    35,W
00FF 0CB5    RRF    35
0100 0CB4    RRF    34
0101 0D35    RLF    35,W
0102 0CB5    RRF    35
0103 0CB4    RRF    34
0104 0D35    RLF    35,W
0105 0CB5    RRF    35
0106 0CB4    RRF    34
0107 0830    MOVF   30,W            if( diff >= step )
0108 022E    SUBWF  2E,W
0109 3080    MOVLW  80h
010A 062F    XORWF  2F,W
010B 00A9    MOVWF  29
010C 3080    MOVLW  80h
010D 0631    XORWF  31,W
010E 1C03    BTFSS  STATUS,C
010F 3E01    ADDLW  01h
0110 0229    SUBWF  29,W
0111 1C03    BTFSS  STATUS,C
0112 2921    GOTO   0121h
0113                                {
0113 1283    BCF    STATUS,RP0         code |= 4;
0114 1539    BSF    39,2
0115 0830    MOVF   30,W               diff -= step;
0116 02AE    SUBWF  2E
0117 0831    MOVF   31,W
0118 1C03    BTFSS  STATUS,C
0119 3E01    ADDLW  01h
011A 02AF    SUBWF  2F
011B 0830    MOVF   30,W               diffq += step;
011C 07B4    ADDWF  34
011D 0831    MOVF   31,W
011E 1803    BTFSC  STATUS,C
011F 3E01    ADDLW  01h
```

```
0120 07B5    ADDWF  35
                                    }
0121 1283    BCF    STATUS,RP0      step >>= 1;
0122 0D31    RLF    31,W
0123 0CB1    RRF    31
0124 0CB0    RRF    30
0125 0830    MOVF   30,W            if( diff >= step )
0126 022E    SUBWF  2E,W
0127 3080    MOVLW  80h
0128 062F    XORWF  2F,W
0129 00A9    MOVWF  29
012A 3080    MOVLW  80h
012B 0631    XORWF  31,W
012C 1C03    BTFSS  STATUS,C
012D 3E01    ADDLW  01h
012E 0229    SUBWF  29,W
012F 1C03    BTFSS  STATUS,C
0130 293F    GOTO   013Fh
0131                                {
0131 1283    BCF    STATUS,RP0         code |= 2;
0132 14B9    BSF    39,1
0133 0830    MOVF   30,W               diff -= step;
0134 02AE    SUBWF  2E
0135 0831    MOVF   31,W
0136 1C03    BTFSS  STATUS,C
0137 3E01    ADDLW  01h
0138 02AF    SUBWF  2F
0139 0830    MOVF   30,W               diffq += step;
013A 07B4    ADDWF  34
013B 0831    MOVF   31,W
013C 1803    BTFSC  STATUS,C
013D 3E01    ADDLW  01h
013E 07B5    ADDWF  35
                                    }
013F 1283    BCF    STATUS,RP0      step >>= 1;
0140 0D31    RLF    31,W
0141 0CB1    RRF    31
0142 0CB0    RRF    30
0143 0830    MOVF   30,W            if( diff >= step )
0144 022E    SUBWF  2E,W
0145 3080    MOVLW  80h
0146 062F    XORWF  2F,W
0147 00A9    MOVWF  29
0148 3080    MOVLW  80h
0149 0631    XORWF  31,W
014A 1C03    BTFSS  STATUS,C
014B 3E01    ADDLW  01h
014C 0229    SUBWF  29,W
014D 1C03    BTFSS  STATUS,C
014E 2957    GOTO   0157h
014F                                {
014F 1283    BCF    STATUS,RP0         code |= 1;
0150 1439    BSF    39,0
0151 0830    MOVF   30,W               diffq += step;
0152 07B4    ADDWF  34
0153 0831    MOVF   31,W
0154 1803    BTFSC  STATUS,C
0155 3E01    ADDLW  01h
0156 07B5    ADDWF  35
                                    }
                                    /* Fixed predictor computes new predicted sample by
                                       adding the old predicted sample to predicted
                                       difference */
0157 1283    BCF    STATUS,RP0      if( code & 8 )
0158 1DB9    BTFSS  39,3
0159 2961    GOTO   0161h
015A 0834    MOVF   34,W               predsample -= diffq;
015B 02B2    SUBWF  32
015C 0835    MOVF   35,W
```



```
015D 1C03    BTFSS  STATUS,C
015E 3E01    ADDLW  01h
015F 02B3    SUBWF  33
0160 2967    GOTO   0167h            else
0161 0834    MOVF   34,W                predsample += diffq;
0162 07B2    ADDWF  32
0163 0835    MOVF   35,W
0164 1803    BTFSC  STATUS,C
0165 3E01    ADDLW  01h
0166 07B3    ADDWF  33
                                    /* Check for overflow of the new predicted sample */
0167 3001    MOVLW  01h              if( predsample > 32767 )
0168 0732    ADDWF  32,W
0169 00A8    MOVWF  28
016A 3080    MOVLW  80h
016B 0633    XORWF  33,W
016C 00A7    MOVWF  27
016D 3000    MOVLW  00h
016E 1803    BTFSC  STATUS,C
016F 3001    MOVLW  01h
0170 0727    ADDWF  27,W
0171 0428    IORWF  28,W
0172 1D03    BTFSS  STATUS,Z
0173 1C03    BTFSS  STATUS,C
0174 297B    GOTO   017Bh
0175 30FF    MOVLW  FFh                 predsample = 32767;
0176 1283    BCF    STATUS,RP0
0177 00B2    MOVWF  32
0178 307F    MOVLW  7Fh
0179 00B3    MOVWF  33
017A 2987    GOTO   0187h            else if( predsample < -32768 )
017B 3080    MOVLW  80h                 predsample = -32768;
017C 1283    BCF    STATUS,RP0
017D 0633    XORWF  33,W
017E 00A7    MOVWF  27
017F 3000    MOVLW  00h
0180 0227    SUBWF  27,W
0181 1803    BTFSC  STATUS,C
0182 2987    GOTO   0187h
0183 1283    BCF    STATUS,RP0
0184 01B2    CLRF   32
0185 3080    MOVLW  80h
0186 00B3    MOVWF  33
                                    /* Find new quantizer stepsize index by adding the
                                     old index to a table lookup using the ADPCM code */
0187 018A    CLRF   PCLATH          index += IndexTable[code];
0188 1283    BCF    STATUS,RP0
0189 0839    MOVF   39,W
018A 2005    CALL   0005h
018B 1283    BCF    STATUS,RP0
018C 07B6    ADDWF  36
                                    /* Check for overflow of the new quantizer step size
                                       index */
018D 1BB6    BTFSC  36,7            if( index < 0 )
018E 01B6    CLRF   36                 index = 0;
018F 3080    MOVLW  80h             if( index > 88 )
0190 1283    BCF    STATUS,RP0
0191 0636    XORWF  36,W
0192 00A7    MOVWF  27
0193 30D8    MOVLW  D8h
0194 0227    SUBWF  27,W
0195 1D03    BTFSS  STATUS,Z
0196 1C03    BTFSS  STATUS,C
0197 299B    GOTO   019Bh
0198 3058    MOVLW  58h                 index = 88;
0199 1283    BCF    STATUS,RP0
019A 00B6    MOVWF  36
                                    /* Save the predicted sample and quantizer step size
                                       index for next iteration */
```

```
019B 1283    BCF    STATUS,RP0        state.prevsample = predsample;
019C 0832    MOVF   32,W
019D 00AB    MOVWF  2B
019E 0833    MOVF   33,W
019F 00AC    MOVWF  2C
01A0 0836    MOVF   36,W              state.previndex = index;
01A1 00AD    MOVWF  2D
                                      /* Return the new ADPCM code */
01A2 300F    MOVLW  0Fh               return ( code & 0x0f );
01A3 0539    ANDWF  39,W
01A4 0008    RETURN
                                  }

/************************************************************************
*    ADPCMDecoder - ADPCM decoder routine                               *
************************************************************************
*    Input Variables:                                                  *
*       char code - 8-bit number containing the 4-bit ADPCM code       *
*    Return Variable:                                                  *
*       signed long - 16-bit signed speech sample                      *
************************************************************************/
                                  signed long ADPCMDecoder(char code )
003A                              {
01A5 1283    BCF    STATUS,RP0
01A6 00BA    MOVWF  3A
                                      /* Restore previous values of predicted sample and
                                         quantizer step size index */
01A7 082B    MOVF   2B,W              predsample = state.prevsample;
01A8 00B2    MOVWF  32
01A9 082C    MOVF   2C,W
01AA 00B3    MOVWF  33
01AB 082D    MOVF   2D,W              index = state.previndex;
01AC 00B6    MOVWF  36
                                      /* Find quantizer step size from lookup table using
                                         index */
01AD 0836    MOVF   36,W              step = StepSizeTable[index];
01AE 0084    MOVWF  FSR
01AF 1003    BCF    STATUS,C
01B0 0D84    RLF    FSR
01B1 3000    MOVLW  00h
01B2 008A    MOVWF  PCLATH
01B3 0804    MOVF   FSR,W
01B4 2016    CALL   0016h
01B5 1283    BCF    STATUS,RP0
01B6 00B0    MOVWF  30
01B7 0A84    INCF   FSR
01B8 3000    MOVLW  00h
01B9 008A    MOVWF  PCLATH
01BA 0804    MOVF   FSR,W
01BB 2016    CALL   0016h
01BC 1283    BCF    STATUS,RP0
01BD 00B1    MOVWF  31
                                      /* Inverse quantize the ADPCM code into a difference
                                         using the quantizer step size */
01BE 0830    MOVF   30,W              diffq = step >> 3;
01BF 00B4    MOVWF  34
01C0 0831    MOVF   31,W
01C1 00B5    MOVWF  35
01C2 0D35    RLF    35,W
01C3 0CB5    RRF    35
01C4 0CB4    RRF    34
01C5 0D35    RLF    35,W
01C6 0CB5    RRF    35
01C7 0CB4    RRF    34
01C8 0D35    RLF    35,W
01C9 0CB5    RRF    35
01CA 0CB4    RRF    34
01CB 1D3A    BTFSS  3A,2              if( code & 4 )
01CC 29D3    GOTO   01D3h
```

```
01CD 0830    MOVF   30,W                diffq += step;
01CE 07B4    ADDWF  34
01CF 0831    MOVF   31,W
01D0 1803    BTFSC  STATUS,C
01D1 3E01    ADDLW  01h
01D2 07B5    ADDWF  35
01D3 1283    BCF    STATUS,RP0        if( code & 2 )
01D4 1CBA    BTFSS  3A,1
01D5 29E3    GOTO   01E3h
01D6 0830    MOVF   30,W                diffq += step >> 1;
01D7 00A8    MOVWF  28
01D8 0831    MOVF   31,W
01D9 00A9    MOVWF  29
01DA 0D29    RLF    29,W
01DB 0CA9    RRF    29
01DC 0CA8    RRF    28
01DD 0828    MOVF   28,W
01DE 07B4    ADDWF  34
01DF 0829    MOVF   29,W
01E0 1803    BTFSC  STATUS,C
01E1 3E01    ADDLW  01h
01E2 07B5    ADDWF  35
01E3 1283    BCF    STATUS,RP0        if( code & 1 )
01E4 1C3A    BTFSS  3A,0
01E5 29F6    GOTO   01F6h
01E6 0830    MOVF   30,W                diffq += step >> 2;
01E7 00A8    MOVWF  28
01E8 0831    MOVF   31,W
01E9 00A9    MOVWF  29
01EA 0D29    RLF    29,W
01EB 0CA9    RRF    29
01EC 0CA8    RRF    28
01ED 0D29    RLF    29,W
01EE 0CA9    RRF    29
01EF 0CA8    RRF    28
01F0 0828    MOVF   28,W
01F1 07B4    ADDWF  34
01F2 0829    MOVF   29,W
01F3 1803    BTFSC  STATUS,C
01F4 3E01    ADDLW  01h
01F5 07B5    ADDWF  35
                                /* Add the difference to the predicted sample */
01F6 1283    BCF    STATUS,RP0        if( code & 8 )
01F7 1DBA    BTFSS  3A,3
01F8 2A00    GOTO   0200h
01F9 0834    MOVF   34,W                predsample -= diffq;
01FA 02B2    SUBWF  32
01FB 0835    MOVF   35,W
01FC 1C03    BTFSS  STATUS,C
01FD 3E01    ADDLW  01h
01FE 02B3    SUBWF  33
01FF 2A06    GOTO   0206h            else
0200 0834    MOVF   34,W                predsample += diffq;
0201 07B2    ADDWF  32
0202 0835    MOVF   35,W
0203 1803    BTFSC  STATUS,C
0204 3E01    ADDLW  01h
0205 07B3    ADDWF  33
                                /* Check for overflow of the new predicted sample */
0206 3001    MOVLW  01h              if( predsample > 32767 )
0207 0732    ADDWF  32,W
0208 00A8    MOVWF  28
0209 3080    MOVLW  80h
020A 0633    XORWF  33,W
020B 00A7    MOVWF  27
020C 3000    MOVLW  00h
020D 1803    BTFSC  STATUS,C
020E 3001    MOVLW  01h
020F 0727    ADDWF  27,W
```

```
0210 0428    IORWF  28,W
0211 1D03    BTFSS  STATUS,Z
0212 1C03    BTFSS  STATUS,C
0213 2A1A    GOTO   021Ah
0214 30FF    MOVLW  FFh                  predsample = 32767;
0215 1283    BCF    STATUS,RP0
0216 00B2    MOVWF  32
0217 307F    MOVLW  7Fh
0218 00B3    MOVWF  33
0219 2A26    GOTO   0226h             else if( predsample < -32768 )
021A 3080    MOVLW  80h                  predsample = -32768;
021B 1283    BCF    STATUS,RP0
021C 0633    XORWF  33,W
021D 00A7    MOVWF  27
021E 3000    MOVLW  00h
021F 0227    SUBWF  27,W
0220 1803    BTFSC  STATUS,C
0221 2A26    GOTO   0226h
0222 1283    BCF    STATUS,RP0
0223 01B2    CLRF   32
0224 3080    MOVLW  80h
0225 00B3    MOVWF  33
                                     /* Find new quantizer step size by adding the old
                                        index and a table lookup using the ADPCM code */
0226 018A    CLRF   PCLATH            index += IndexTable[code];
0227 1283    BCF    STATUS,RP0
0228 083A    MOVF   3A,W
0229 2005    CALL   0005h
022A 1283    BCF    STATUS,RP0
022B 07B6    ADDWF  36
                                     /* Check for overflow of the new quantizer step size
                                        index */
022C 1BB6    BTFSC  36,7              if( index < 0 )
022D 01B6    CLRF   36                   index = 0;
022E 3080    MOVLW  80h               if( index > 88 )
022F 1283    BCF    STATUS,RP0
0230 0636    XORWF  36,W
0231 00A7    MOVWF  27
0232 30D8    MOVLW  D8h
0233 0227    SUBWF  27,W
0234 1D03    BTFSS  STATUS,Z
0235 1C03    BTFSS  STATUS,C
0236 2A3A    GOTO   023Ah
0237 3058    MOVLW  58h                  index = 88;
0238 1283    BCF    STATUS,RP0
0239 00B6    MOVWF  36
                                     /* Save predicted sample and quantizer step size
                                        index for next iteration */
023A 1283    BCF    STATUS,RP0        state.prevsample = (short)predsample;
023B 0832    MOVF   32,W
023C 00AB    MOVWF  2B
023D 01AC    CLRF   2C
023E 1BAB    BTFSC  2B,7
023F 03AC    DECF   2C
0240 1283    BCF    STATUS,RP0        state.previndex = index;
0241 0836    MOVF   36,W
0242 00AD    MOVWF  2D
                                     /* Return the new speech sample */
0243 0833    MOVF   33,W              return( predsample );
0244 0084    MOVWF  FSR
0245 0832    MOVF   32,W
0246 0008    RETURN
                                   }
```

APPENDIX D: INTERACTIVE MULTIMEDIA ASSOCIATION INFORMATION

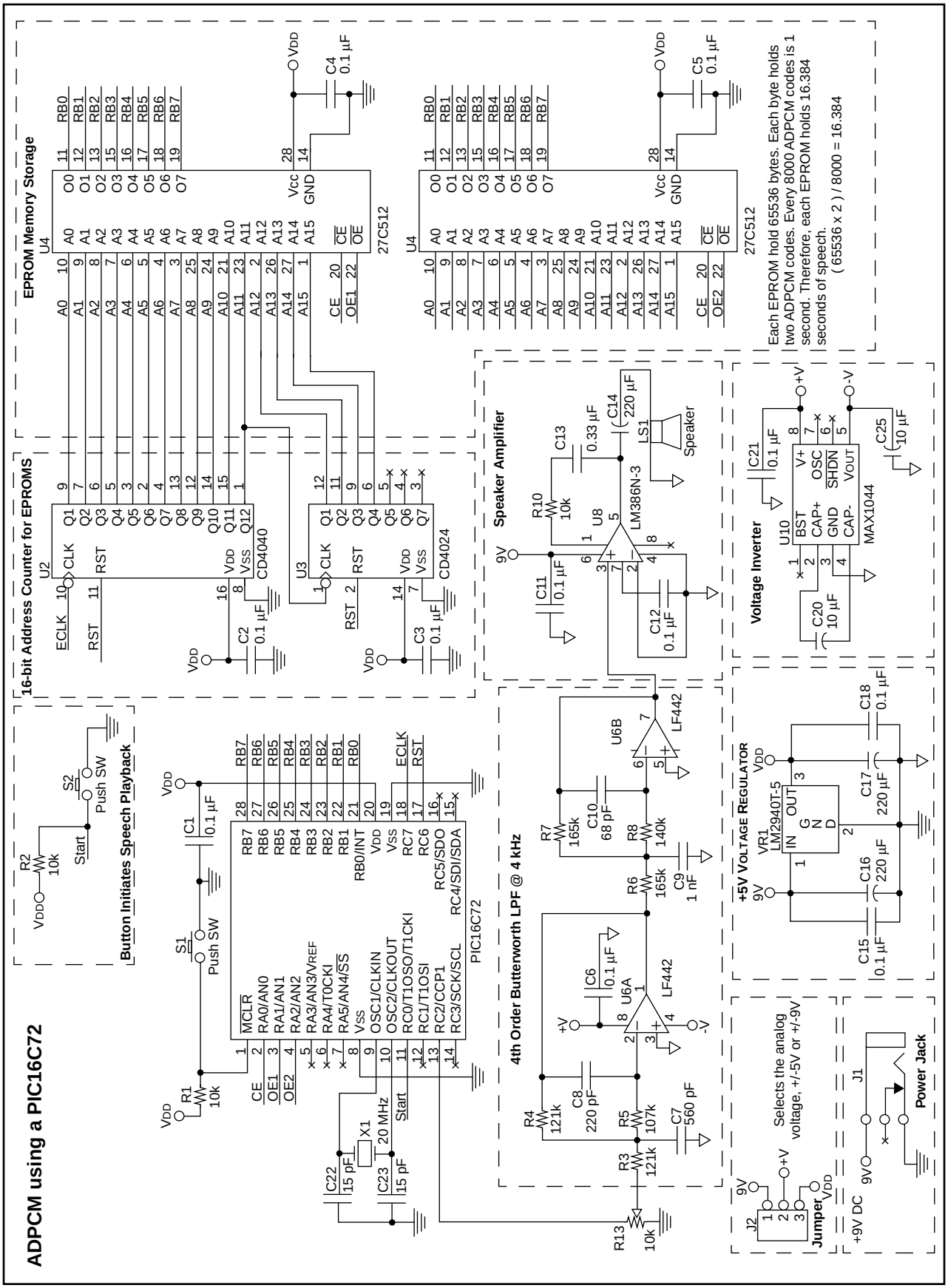
The IMA ADPCM Reference Algorithm is contained in the Digital Audio Doc-Pac. The Doc-Pac contains the following information:

- *About IMA Digital Audio* - describes the IMA's activities in digital audio
- IMA *Recommended Practices for Enhancing Digital Audio Compatability in Multimedia Systems* (version 3.0). This document contains the official technical recommendations including the ADPCM algorithm
- *IMA Digital Audio Special Edition Proceedings* This document contains a detailed notes of the evaluation process, code fragments and testing methodologies.
- Floppy disk with code fragments.
- Contact information for companies supporting IMA ADPCM.

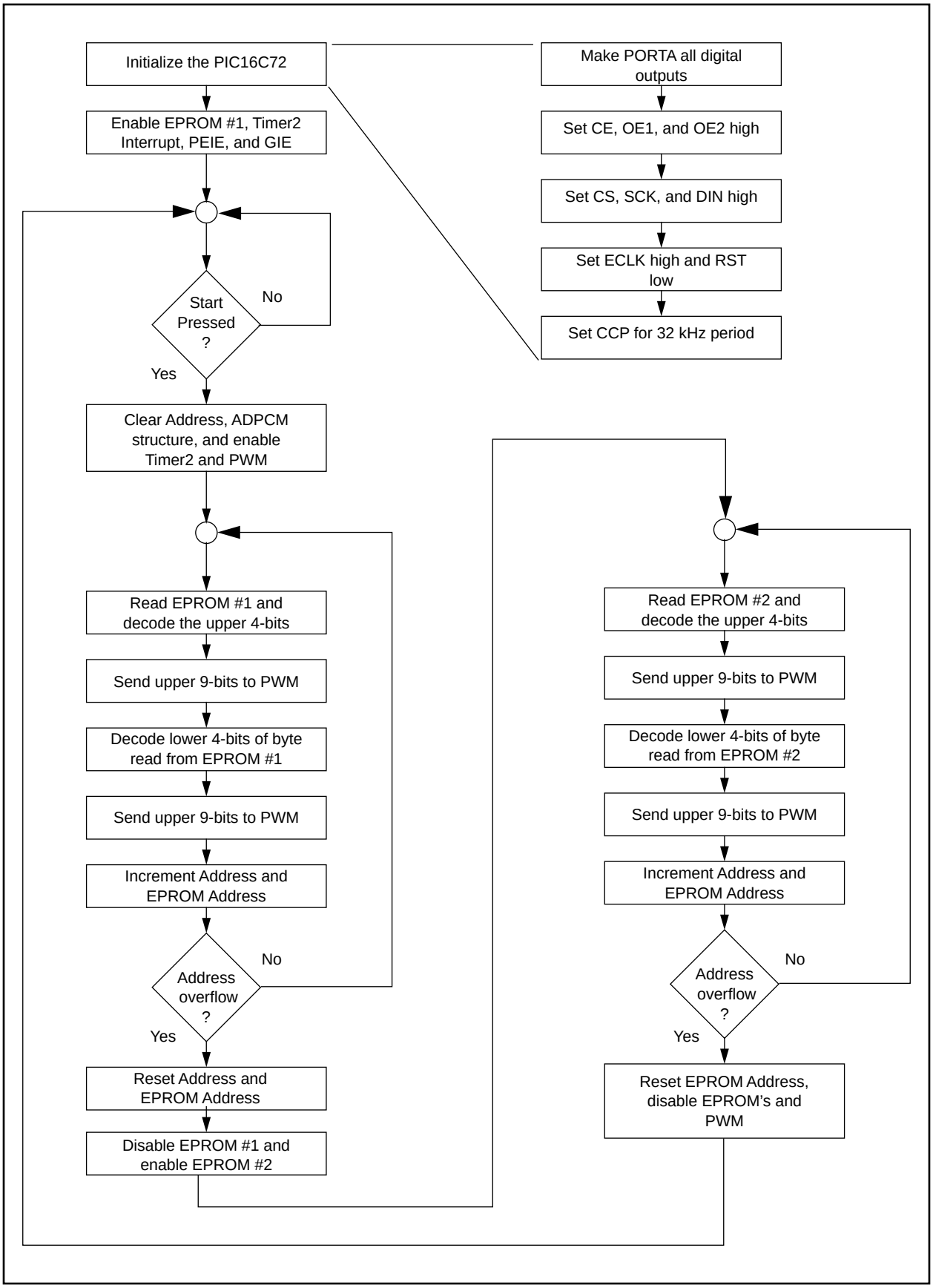
The IMA can be reached at:

Interactive Multimedia Association
48 Maryland Avenue, Suite 202
Annapolis, MD 21401-8011 USA
Tel: (410)-626-1380
Fax: (410)-263-0590
http://www.ima.org

APPENDIX E: APPLICATION HARDWARE SCHEMATIC



APPENDIX F: APPLICATION FIRMWARE BLOCK DIAGRAM AND LISTING



Firmware Listing

```
/*****************************************************************************
*    Filename:   SPEECH.C                                                    *
*****************************************************************************
*    Author:     Rodger Richey                                               *
*    Title:      Senior Applications Engineer                                *
*    Company:    Microchip Technology Incorporated                           *
*    Revision:   0                                                           *
*    Date:       1-9-96                                                      *
*    Compiled using Bytecraft Ltd. MPC version BC.193                        *
*****************************************************************************
*    Include files:                                                          *
*       16c72.h - Standard include file for the PIC16C72 device              *
*       delay14.h - Standard include file for delay routines                 *
*       adpcm.h - Include file with ADPCM related information (Rev0)         *
*       adpcm.c - Source file with ADPCM routines (Rev0)                     *
*****************************************************************************
*    This file contains the routines to:                                     *
*       - Initialize the PIC16C72                                            *
*       - Read data from the EPROMs                                          *
*       - Call the ADPCM decoder routine                                     *
*       - Send data to the DAC and PWM                                       *
*       - Increment the address of the EPROMs                                *
*****************************************************************************
*    External Clock Frequency: 20MHz                                         *
*       Configuration Bit Settings                                           *
*            OSC:    HS                                                      *
*            WDT:    Off                                                     *
*            PWRTE:  On                                                      *
*            CP:     Off                                                     *
*    Compression and Decompression (only Decompression is used)              *
*       Program Memory Usage:          906 words                             *
*       Data Memory Usage:             24 bytes                              *
*    Compression only                                                        *
*       Program Memory Usage:          686 words                             *
*       Data Memory Usage:             22 bytes                              *
*****************************************************************************/
                                #pragma option v
                                #include <16c72.h>
                                #pragma option +l;
0FFB                            #define MAXROM 2043
0005 0FFB                       #pragma memory ROM [MAXROM] @ 0x05;
0020 0060                       #pragma memory RAM [0x96]   @ 0x20;
00A0 0060                       #pragma memory RAM [0x32]   @ 0xA0;
                                #pragma option +l;

                                #include "adpcm.h"
/*****************************************************************************
*    Filename:   PCADPCM.H                                                   *
*****************************************************************************
*    Author:     Rodger Richey                                               *
*    Title:      Senior Applications Engineer                                *
*    Company:    Microchip Technology Incorporated                           *
*    Revision:   0                                                           *
*    Date:       1-11-96                                                     *
*    Compiled using Bytecraft Ltd. MPC version BC.193                        *
*****************************************************************************
*    This is the header file that contains the ADPCM structure definition    *
*    and the function prototypes.                                            *
*****************************************************************************/
                                struct ADPCMState {
                                   signed long prevsample;       / * Predicted sample */
                                   int         previndex; /*Index into step size table*/
                                };
```

```
                                /* Function prototype for the ADPCM Encoder routine */
                                signed long ADPCMDecoder( char );

                                /* Function prototype for the ADPCM Decoder routine */
                                char ADPCMEncoder( signed long );

                                // Defines for PORTA
0000                            #define CE   0   // Chip Enable for EPROMS
0001                            #define OE1  1   // Output Enable for EPROM #1
0002                            #define OE2  2   // Output Enable for EPROM #2

                                // Defines for PORTC
0000                            #define STRT 0   // Start Button input
0002                            #define PWM  2   // PWM output
0006                            #define RST  6   // EPROM counter reset
0007                            #define ECLK 7   // EPROM counter clock

0025                            unsigned long Address;  // EPROM address
002A                            unsigned char Wait;
002B 0003                       struct ADPCMState state;// ADPCM structure

                                #include "adpcm.c"
```

----------Insert Appendix C here----------

```
/*************************************************************************
*    Init72 - This function initializes PORTA, PORTC, and PWM            *
*************************************************************************
*    Input Variables:                                                    *
*       None                                                             *
*    Return Variables:                                                   *
*       None                                                             *
*************************************************************************/
                                void Init72(void)
                                {
0247 3008    MOVLW  08h             OPTION = 8;
0248 1683    BSF    STATUS,RP0
0249 0081    MOVWF  TMR0
024A 3007    MOVLW  07h             ADCON1 = 7;    // Make PORTA all digital
024B 009F    MOVWF  ADCON0
024C 1283    BCF    STATUS,RP0      PORTA = 7;     // Set CE,OE1,OE2
024D 0085    MOVWF  PORTA
024E 1683    BSF    STATUS,RP0      TRISA = 0;     // Make PORTA all outputs
024F 0185    CLRF   PORTA
0250 3040    MOVLW  40h             PORTC = 0x40;  // Set ECLK, all outputs
0251 1283    BCF    STATUS,RP0
0252 0087    MOVWF  PORTC
0253 3005    MOVLW  05h             TRISC = 0x05;  // Clear RST output, STRT input
0254 1683    BSF    STATUS,RP0
0255 0087    MOVWF  PORTC
0256 309B    MOVLW  9Bh             PR2 = 155;     // Set PWM for 32kHz period
0257 0092    MOVWF  T2CON
0258 1283    BCF    STATUS,RP0      CCPR1L = 0;
0259 0195    CLRF   CCPR1L
025A 3018    MOVLW  18h             T2CON = 0x18;  // Set Timer2 postscaler to 4
025B 0092    MOVWF  T2CON
025C 0008    RETURN                 return;
                                }

/*************************************************************************
*    main - This function controls everything, kinda like a god.         *
*************************************************************************
*    Input Variables:                                                    *
```

```
*       None                                                                  *
*    Return Variables:                                                        *
*       None                                                                  *
************************************************************************/
                                void main(void)
                                {
003B                                char code;              // EPROM byte
003C                                signed long sample;     // decoded sample
003E                                unsigned long pwmout;  // temporary variable for PWM

025D 2247    CALL   0247h           Init72();               // Initialize the PIC16C72

025E 3001    MOVLW  01h             Wait = 1;
025F 1283    BCF    STATUS,RP0
0260 00AA    MOVWF  2A
0261 1707    BSF    PORTC,6         PORTC.RST = 1;          // Reset the EPROM counter
0262 0000    NOP                    NOP();
0263 0000    NOP                    NOP();
0264 1307    BCF    PORTC,6         PORTC.RST = 0;
0265 0000    NOP                    NOP();
0266 0000    NOP                    NOP();
0267 1085    BCF    PORTA,1         PORTA.OE1 = 0;          // Enable EPROM #1
0268 0000    NOP                    NOP();
0269 0000    NOP                    NOP();
026A 1005    BCF    PORTA,0         PORTA.CE = 0;
026B 0000    NOP                    NOP();
026C 0000    NOP                    NOP();
026D 108C    BCF    PIR1,1          PIR1.TMR2IF = 0;        // Enable Timer2 Interrupts
026E 1683    BSF    STATUS,RP0      PIE1.TMR2IE = 1;
026F 148C    BSF    PIR1,1
0270 170B    BSF    INTCON,PEIE     INTCON.PEIE = 1;        // Enable PEIE and GIE
0271 178B    BSF    INTCON,GIE      INTCON.GIE = 1;

                                    while(1)
                                    {
0272 1283    BCF    STATUS,RP0
0273 1807    BTFSC  PORTC,0
0274 2A72    GOTO   0272h
0275                                   while(PORTC.STRT); // Wait for Start button to be
0275 1283    BCF    STATUS,RP0         while(!PORTC.STRT); // pressed then released
0276 1C07    BTFSS  PORTC,0
0277 2A75    GOTO   0275h
0278 1283    BCF    STATUS,RP0         Address = 0;        // Clear EPROM address
0279 01A5    CLRF   25
027A 01A6    CLRF   26
027B 01AB    CLRF   2B                 state.prevsample = 0; // Clear ADPCM structure
027C 01AC    CLRF   2C
027D 01AD    CLRF   2D                 state.previndex = 0;

027E 1683    BSF    STATUS,RP0         TRISC.PWM = 0;      // Make PWM an output
027F 1107    BCF    PORTC,2
0280 1283    BCF    STATUS,RP0         T2CON.TMR2ON = 1;   // Enable Timer2
0281 1512    BSF    T2CON,2
0282 300F    MOVLW  0Fh                CCP1CON = 0x0f;     // Enable PWM
0283 0097    MOVWF  CCP1CON
                                       do                  // Decode upper 4 bits of code
                                       {
0284 1283    BCF    STATUS,RP0           code = PORTB;
0285 0806    MOVF   PORTB,W
0286 00BB    MOVWF  3B
0287 083B    MOVF   3B,W                 sample = ADPCMDecoder( (code>>4)&0x0f );
0288 00A7    MOVWF  27
0289 0EA7    SWAPF  27
028A 300F    MOVLW  0Fh
028B 0527    ANDWF  27,W
028C 390F    ANDLW  0Fh
```

```
028D 21A5    CALL   01A5h
028E 1283    BCF    STATUS,RP0
028F 00BC    MOVWF  3C
0290 0804    MOVF   FSR,W
0291 00BD    MOVWF  3D
0292 1283    BCF    STATUS,RP0            while(Wait);     // Wait for 8kHz to output
0293 082A    MOVF   2A,W
0294 3800    IORLW  00h
0295 1D03    BTFSS  STATUS,Z
0296 2A92    GOTO   0292h
0297 3001    MOVLW  01h                   Wait = 1;
0298 1283    BCF    STATUS,RP0
0299 00AA    MOVWF  2A
029A 083C    MOVF   3C,W                  pwmout = sample; // Write to PWM
029B 00BE    MOVWF  3E
029C 083D    MOVF   3D,W
029D 00BF    MOVWF  3F
029E 1FBD    BTFSS  3D,7                  if(sample<0)     // Add offset to
029F 2AA4    GOTO   02A4h
02A0 3080    MOVLW  80h                    pwmout -= 0x8000; // sample
02A1 023F    SUBWF  3F,W
02A2 00BF    MOVWF  3F
02A3 2AA6    GOTO   02A6h                 else
02A4 3080    MOVLW  80h                    pwmout += 0x8000;
02A5 07BF    ADDWF  3F
02A6 083F    MOVF   3F,W                  CCPR1L = (pwmout>>9)&0x007f; // Write 7 bits
02A7 00A7    MOVWF  27
02A8 0CA7    RRF    27
02A9 307F    MOVLW  7Fh
02AA 0527    ANDWF  27,W
02AB 0095    MOVWF  CCPR1L
02AC 30CF    MOVLW  CFh                   CCP1CON &= 0xcf;             // to CCPR1L
02AD 0597    ANDWF  CCP1CON
02AE 183F    BTFSC  3F,0                  if(pwmout&0x0100)            // Write 2 bits
02AF 1697    BSF    CCP1CON,5              CCP1CON.5 = 1;              // to CCP1CON
02B0 1283    BCF    STATUS,RP0            if(pwmout&0x0080)            // to get 9-bit
02B1 1BBE    BTFSC  3E,7
02B2 1617    BSF    CCP1CON,4              CCP1CON.4 = 1;              // PWM

                                                      // Decode lower 4 bits of code
02B3 300F    MOVLW  0Fh                   sample = ADPCMDecoder(code&0x0f);
02B4 1283    BCF    STATUS,RP0
02B5 053B    ANDWF  3B,W
02B6 118A    BCF    PCLATH,3
02B7 21A5    CALL   01A5h
02B8 118A    BCF    PCLATH,3
02B9 1283    BCF    STATUS,RP0
02BA 00BC    MOVWF  3C
02BB 0804    MOVF   FSR,W
02BC 00BD    MOVWF  3D
02BD 1283    BCF    STATUS,RP0            while(Wait);     // Wait for 8kHz to output
02BE 082A    MOVF   2A,W
02BF 3800    IORLW  00h
02C0 1D03    BTFSS  STATUS,Z
02C1 2ABD    GOTO   02BDh
02C2 3001    MOVLW  01h                   Wait = 1;
02C3 1283    BCF    STATUS,RP0
02C4 00AA    MOVWF  2A
02C5 083C    MOVF   3C,W                  pwmout = sample; // Write to PWM
02C6 00BE    MOVWF  3E
02C7 083D    MOVF   3D,W
02C8 00BF    MOVWF  3F
02C9 1FBD    BTFSS  3D,7                  if(sample<0)     // Add offset to
02CA 2ACF    GOTO   02CFh
02CB 3080    MOVLW  80h                    pwmout -= 0x8000; // sample
02CC 023F    SUBWF  3F,W
```

```
02CD 00BF    MOVWF  3F
02CE 2AD1    GOTO   02D1h                 else
02CF 3080    MOVLW  80h                   pwmout += 0x8000;
02D0 07BF    ADDWF  3F
02D1 083F    MOVF   3F,W                 CCPR1L = (pwmout>>9)&0x007f; // Write 7 bits
02D2 00A7    MOVWF  27
02D3 0CA7    RRF    27
02D4 307F    MOVLW  7Fh
02D5 0527    ANDWF  27,W
02D6 0095    MOVWF  CCPR1L
02D7 30CF    MOVLW  CFh                  CCP1CON &= 0xcf;             // to CCPR1L
02D8 0597    ANDWF  CCP1CON
02D9 183F    BTFSC  3F,0                 if(pwmout&0x0100)            // Write 2 bits
02DA 1697    BSF    CCP1CON,5             CCP1CON.5 = 1;              // to CCP1CON
02DB 1283    BCF    STATUS,RP0           if(pwmout&0x0080)            // to get 9-bit
02DC 1BBE    BTFSC  3E,7
02DD 1617    BSF    CCP1CON,4             CCP1CON.4 = 1;              // PWM

                                                      // Increment Address of EPROMS
02DE 1283    BCF    STATUS,RP0           Address++;
02DF 0AA5    INCF   25
02E0 1903    BTFSC  STATUS,Z
02E1 0AA6    INCF   26
02E2 1787    BSF    PORTC,7              PORTC.ECLK = 1;
02E3 0000    NOP                         NOP();
02E4 0000    NOP                         NOP();
02E5 1387    BCF    PORTC,7              PORTC.ECLK = 0;
02E6 0000    NOP                         NOP();
02E7 0000    NOP                         NOP();
02E8 0826    MOVF   26,W                } while(Address);  // Has Address overflowed?
02E9 0425    IORWF  25,W
02EA 118A    BCF    PCLATH,3
02EB 1D03    BTFSS  STATUS,Z
02EC 2A84    GOTO   0284h

02ED 1283    BCF    STATUS,RP0          PORTA.CE = 1;      // Disable EPROM #1
02EE 1405    BSF    PORTA,0
02EF 0000    NOP                        NOP();
02F0 0000    NOP                        NOP();
02F1 1485    BSF    PORTA,1             PORTA.OE1 = 1;
02F2 0000    NOP                        NOP();
02F3 0000    NOP                        NOP();
02F4 1705    BSF    PORTA,6             PORTA.RST = 1;     // Reset EPROM counter
02F5 0000    NOP                        NOP();
02F6 0000    NOP                        NOP();
02F7 1305    BCF    PORTA,6             PORTA.RST = 0;
02F8 0000    NOP                        NOP();
02F9 0000    NOP                        NOP();
02FA 1105    BCF    PORTA,2             PORTA.OE2 = 0;     // Enable EPROM #2
02FB 0000    NOP                        NOP();
02FC 0000    NOP                        NOP();
02FD 1005    BCF    PORTA,0             PORTA.CE = 0;
02FE 0000    NOP                        NOP();
02FF 0000    NOP                        NOP();
0300 01A5    CLRF   25                  Address = 0;       // Clear EPROM Address
0301 01A6    CLRF   26
0302 0000    NOP                        NOP();
0303 0000    NOP                        NOP();

                                        do
                                        {                  // Decode upper 4 bits of code
0304 1283    BCF    STATUS,RP0            code = PORTB;
0305 0806    MOVF   PORTB,W
0306 00BB    MOVWF  3B
0307 083B    MOVF   3B,W                  sample = ADPCMDecoder( (code>>4)&0x0f );
0308 00A7    MOVWF  27
```

```
0309 0EA7    SWAPF  27
030A 300F    MOVLW  0Fh
030B 0527    ANDWF  27,W
030C 390F    ANDLW  0Fh
030D 21A5    CALL   01A5h
030E 1283    BCF    STATUS,RP0
030F 00BC    MOVWF  3C
0310 0804    MOVF   FSR,W
0311 00BD    MOVWF  3D
0312 1283    BCF    STATUS,RP0            while(Wait);     // Wait for 8kHz
0313 082A    MOVF   2A,W
0314 3800    IORLW  00h
0315 1D03    BTFSS  STATUS,Z
0316 2B12    GOTO   0312h
0317 3001    MOVLW  01h                   Wait = 1;
0318 1283    BCF    STATUS,RP0
0319 00AA    MOVWF  2A
031A 083C    MOVF   3C,W                  pwmout = sample; // Write to PWM
031B 00BE    MOVWF  3E
031C 083D    MOVF   3D,W
031D 00BF    MOVWF  3F
031E 1FBD    BTFSS  3D,7                  if(sample<0)     // Add offset to
031F 2B24    GOTO   0324h
0320 3080    MOVLW  80h                    pwmout -= 0x8000; // sample
0321 023F    SUBWF  3F,W
0322 00BF    MOVWF  3F
0323 2B26    GOTO   0326h                 else
0324 3080    MOVLW  80h                    pwmout += 0x8000;
0325 07BF    ADDWF  3F
0326 083F    MOVF   3F,W                  CCPR1L = (pwmout>>9)&0x007f; // Write 7 bits
0327 00A7    MOVWF  27
0328 0CA7    RRF    27
0329 307F    MOVLW  7Fh
032A 0527    ANDWF  27,W
032B 0095    MOVWF  CCPR1L
032C 30CF    MOVLW  CFh                   CCP1CON &= 0xcf;             // to CCPR1L
032D 0597    ANDWF  CCP1CON
032E 183F    BTFSC  3F,0                  if(pwmout&0x0100)            // Write 2 bits
032F 1697    BSF    CCP1CON,5              CCP1CON.5 = 1;              // to CCP1CON
0330 1283    BCF    STATUS,RP0            if(pwmout&0x0080)            // to get 9-bit
0331 1BBE    BTFSC  3E,7
0332 1617    BSF    CCP1CON,4              CCP1CON.4 = 1;              // PWM

                                                    // Decode lower 4 bits of code
0333 300F    MOVLW  0Fh                   sample = ADPCMDecoder(code&0x0f);
0334 1283    BCF    STATUS,RP0
0335 053B    ANDWF  3B,W
0336 118A    BCF    PCLATH,3
0337 21A5    CALL   01A5h
0338 118A    BCF    PCLATH,3
0339 1283    BCF    STATUS,RP0
033A 00BC    MOVWF  3C
033B 0804    MOVF   FSR,W
033C 00BD    MOVWF  3D
033D 1283    BCF    STATUS,RP0            while(Wait);     // Wait for 8kHz
033E 082A    MOVF   2A,W
033F 3800    IORLW  00h
0340 1D03    BTFSS  STATUS,Z
0341 2B3D    GOTO   033Dh
0342 3001    MOVLW  01h                   Wait = 1;
0343 1283    BCF    STATUS,RP0
0344 00AA    MOVWF  2A
0345 083C    MOVF   3C,W                  pwmout = sample; // Write to PWM
0346 00BE    MOVWF  3E
0347 083D    MOVF   3D,W
0348 00BF    MOVWF  3F
```

```
0349 1FBD    BTFSS  3D,7                  if(sample<0)     // Add offset to
034A 2B4F    GOTO   034Fh
034B 3080    MOVLW  80h                    pwmout -= 0x8000; // sample
034C 023F    SUBWF  3F,W
034D 00BF    MOVWF  3F
034E 2B51    GOTO   0351h                  else
034F 3080    MOVLW  80h                    pwmout += 0x8000;
0350 07BF    ADDWF  3F
0351 083F    MOVF   3F,W                   CCPR1L = (pwmout>>9)&0x007f; // Write 7 bits
0352 00A7    MOVWF  27
0353 0CA7    RRF    27
0354 307F    MOVLW  7Fh
0355 0527    ANDWF  27,W
0356 0095    MOVWF  CCPR1L
0357 30CF    MOVLW  CFh                    CCP1CON &= 0xcf;             // to CCPR1L
0358 0597    ANDWF  CCP1CON
0359 183F    BTFSC  3F,0                   if(pwmout&0x0100)            // Write 2 bits
035A 1697    BSF    CCP1CON,5               CCP1CON.5 = 1;              // to CCP1CON
035B 1283    BCF    STATUS,RP0             if(pwmout&0x0080)            // to get 9-bit
035C 1BBE    BTFSC  3E,7
035D 1617    BSF    CCP1CON,4               CCP1CON.4 = 1;              // PWM

                                                                 // Increment Address of EPROMS
035E 1283    BCF    STATUS,RP0             Address++;
035F 0AA5    INCF   25
0360 1903    BTFSC  STATUS,Z
0361 0AA6    INCF   26
0362 1787    BSF    PORTC,7                PORTC.ECLK = 1;
0363 0000    NOP                           NOP();
0364 0000    NOP                           NOP();
0365 1387    BCF    PORTC,7                PORTC.ECLK = 0;
0366 0000    NOP                           NOP();
0367 0000    NOP                           NOP();
0368 0826    MOVF   26,W                  } while(Address);  // Has Address overflowed?
0369 0425    IORWF  25,W
036A 118A    BCF    PCLATH,3
036B 1D03    BTFSS  STATUS,Z
036C 2B04    GOTO   0304h
036D 1283    BCF    STATUS,RP0           T2CON.TMR2ON = 0;  // Disable Timer2
036E 1112    BCF    T2CON,2
036F 0197    CLRF   CCP1CON              CCP1CON = 0;       // Disable PWM
0370 1683    BSF    STATUS,RP0           TRISC.PWM = 1;     // Make PWM input
0371 1507    BSF    PORTC,2
0372 1283    BCF    STATUS,RP0           PORTA.CE = 1;      // Disable EPROM #2
0373 1405    BSF    PORTA,0
0374 0000    NOP                         NOP();
0375 0000    NOP                         NOP();
0376 1505    BSF    PORTA,2              PORTA.OE2 = 1;
0377 0000    NOP                         NOP();
0378 0000    NOP                         NOP();
0379 1705    BSF    PORTA,6              PORTA.RST = 1;     // Reset EPROM counter
037A 0000    NOP                         NOP();
037B 0000    NOP                         NOP();
037C 1305    BCF    PORTA,6              PORTA.RST = 0;
037D 0000    NOP                         NOP();
037E 0000    NOP                         NOP();
037F 1085    BCF    PORTA,1              PORTA.OE1 = 0;     // Enable EPROM #1
0380 0000    NOP                         NOP();
0381 0000    NOP                         NOP();
0382 1005    BCF    PORTA,0              PORTA.CE = 0;
0383 2A72    GOTO   0272h              }
0384 0008    RETURN                 }

/************************************************************************
*    __INT - This is the interrupt service routine. Only Timer 2 overflow    *
*            is implemented.                                                 *
```

```
*************************************************************************
*    Input Variables:                                                   *
*       None                                                            *
*    Return Variables:                                                  *
*       None                                                            *
*************************************************************************/
0004 2B85    GOTO   0385h         void __INT(void)
0385                              {
0385 1283    BCF    STATUS,RP0        if(PIR1.TMR2IF)       // Timer2 overflow interrupt
0386 1C8C    BTFSS  PIR1,1
0387 2B8A    GOTO   038Ah
0388                                   {
0388 01AA    CLRF   2A                    Wait = 0;
0389 108C    BCF    PIR1,1                PIR1.TMR2IF = 0;   // Clear flag
                                       }
038A 0009    RETFIE                    return;
                                  }
ROM USAGE MAP

    0000 to 0002    0004 to 038A
    Total ROM used 038A

Errors             :    0
Warnings           :    0
```

APPENDIX G: PC ADPCM ENCODER/DECODER PROGRAM

PCSPEECH.C

```
/*************************************************************************
*       Filename:  PCSPEECH.C                                            *
*************************************************************************
*       Author:         Rodger Richey                                    *
*       Title:         Senior Applications Engineer                      *
*       Company:        Microchip Technology Incorporated                *
*       Revision:       0                                                *
*       Date:         1-11-96                                            *
*       Compiled using Borland C+ Version 3.1                            *
*************************************************************************
*       Include files:                                                   *
*               stdio.h - Standard input/output header file              *
*               stdlib.h - Standard library header file                  *
*               string.h - Standard string header file                   *
*               pcadpcm.h - ADPCM related information header file (Rev0) *
*************************************************************************
*       Usage:                                                           *
*               ADPCM Encode - pcspeech e <infile> <outfile>             *
*                              <infile> is a 16-bit raw speech file      *
*                              <outfile> contains the ADPCM codes        *
*               ADPCM Decode - pcspeech d <infile> <outfile>             *
*                              <infile> contains the ADPCM codes         *
*                              <outfile> is a 16-bit raw speech file     *
*************************************************************************
*       This file contains the code to:                                  *
*               - Open the input and output files                        *
*               - Read data from the input file                          *
*               - Call the encode/decode routines                        *
*               - Write data to the output file                          *
*************************************************************************/
#include <stdio.h>
#include <stdlib.h>
#include <string.h>
#include "pcadpcm.h"

/*************************************************************************
*       Usage - this routine prints a how to message for the pcspeech prgm *
*************************************************************************
*       Input variables:                                                 *
*               None                                                     *
*       Output variables:                                                *
*               None                                                     *
*************************************************************************/
void Usage(void)
{
    printf("ADPCM Encoder/Decoder -- usage:\n");
    printf("\tEncoder = pcspeech e infile outfile\n");
    printf("\tDecoder = pcspeech d infile outfile\n");
    exit(1);
}

/*************************************************************************
*       main - controls file I/O and ADPCM calls                         *
*************************************************************************
*       Input variables:                                                 *
*               int argc - number of arguements in argv                  *
*               char **argv - pointer to an array of strings             *
*       Return variables:                                                *
*               None                                                     *
*************************************************************************/
void main(
    int   argc,
    char  **argv)
```

```
{
    int         which;
    short                   sample;
    unsigned char           code;
    int                     n;
    struct ADPCMstate       state;
    FILE        *fpin;
    FILE        *fpout;

    state.prevsample=0;
    state.previndex=0;

    /* Determine if this is an encode or decode operation */
    if(argc <= 1)
        Usage();
    else if( strcmp(argv[1],"e")==0 || strcmp(argv[1],"E")==0 )
        which = 0;
    else if( strcmp(argv[1],"d")==0 || strcmp(argv[1],"D")==0 )
        which = 1;
    argc--;
    argv++;

    /* Open input file for processing */
    if(argc <= 1)
        Usage();
    else if( (fpin=fopen(argv[1],"rb"))==NULL)
    {
        printf("ADPCM Encoder/Decoder\n");
        printf("ERROR: Could not open %s for input\n",argv[1]);
        exit(1);
    }
    argc--;
    argv++;

    /* Open output file */
    if(argc <= 1)
    {
        fclose(fpin);
        Usage();
    }
    else if( (fpout=fopen(argv[1],"wb"))==NULL)
    {
        fclose(fpin);
        printf("ADPCM Encoder/Decoder\n");
        printf("ERROR: Could not open %s for output\n",argv[1]);
        exit(1);
    }

    // ADPCM Decoder selected
    if(which)
    {
        printf("ADPCM Decoding in progress\n");
        /* Read and unpack input codes and process them */
        while (fread(&code, sizeof (char), 1, fpin) == 1)
        {
            // Send the upper 4-bits of code to decoder
            sample = ADPCMDecoder((code>>4)&0x0f, &state);
            // Write sample for upper 4-bits of code
            fwrite(&sample, sizeof(short), 1, fpout);

            // Send the lower 4-bits of code to decoder
            sample = ADPCMDecoder(code&0x0f,&state);
            // Write sample for lower 4-bits of code
            fwrite(&sample,sizeof(short),1,fpout);
        }
    }
```

```
    // ADPCM Encoder selected
    else
    {
        printf("ADPCM Encoding in progress\n");
        /* Read input file and process */
        while (fread(&sample, sizeof(short), 1, fpin) == 1)
        {
            // Encode sample into lower 4-bits of code
            code = ADPCMEncoder(sample,&state);
            // Move ADPCM code to upper 4-bits
            code = (code << 4) & 0xf0;
            // Read new sample from file
            if(fread(&sample,sizeof(short),1,fpin)==0)
            {
                // No more samples, write code to file
                fwrite(&code,sizeof(char),1,fpout);
                break;
            }
            // Encode sample and save in lower 4-bits of code
            code |= ADPCMEncoder(sample,&state);
            // Write code to file, code contains 2 ADPCM codes
            fwrite(&code, sizeof (char), 1, fpout);
        }
    }

    fclose(fpin);
    fclose(fpout);
}
```

PCADPCM.H

```
/*************************************************************************
*       Filename:  PCADPCM.H                                             *
*************************************************************************
*       Author:         Rodger Richey                                    *
*       Title:         Senior Applications Engineer                     *
*       Company:        Microchip Technology Incorporated                *
*       Revision:       0                                                *
*       Date:         1-11-96                                           *
*       Compiled using Borland C+ Version 3.1                            *
*************************************************************************
*       This is the header file that contains the ADPCM structure definition *
*       and the function prototypes.                                     *
*************************************************************************/
struct ADPCMstate {
    short prevsample;/* Predicted sample */
    int previndex;/* Index into step size table */
};

/* Function prototype for the ADPCM Encoder routine */
char ADPCMEncoder(short , struct ADPCMstate *);

/* Function prototype for the ADPCM Decoder routine */
int ADPCMDecoder(char , struct ADPCMstate *);
```

PCADPCM.C

```
/*************************************************************************
*       Filename:  PCADPCM.C                                             *
*************************************************************************
*       Author:         Rodger Richey                                    *
*       Title:         Senior Applications Engineer                     *
*       Company:        Microchip Technology Incorporated                *
*       Revision:       0                                                *
*       Date:         1-11-96                                           *
*       Compiled using Borland C+ Version 3.1                            *
```

```
*************************************************************************
*       Include files:                                                  *
*               stdio.h - Standard input/output header file             *
*               pcadpcm.h - ADPCM related information header file (Rev0)*
*************************************************************************
*       This file contains the ADPCM encode and decode routines.  These *
*       routines were obtained from the Interactive Multimedia Association's *
*       Reference ADPCM algorithm.  This algorithm was first implemented by  *
*       Intel/DVI.                                                      *
*************************************************************************/

#include <stdio.h>
#include "pcadpcm.h"

/* Table of index changes */
signed char IndexTable[16] = {
    -1, -1, -1, -1, 2, 4, 6, 8,
    -1, -1, -1, -1, 2, 4, 6, 8,
};

/* Quantizer step size lookup table */
int StepSizeTable[89] = {
    7, 8, 9, 10, 11, 12, 13, 14, 16, 17,
    19, 21, 23, 25, 28, 31, 34, 37, 41, 45,
    50, 55, 60, 66, 73, 80, 88, 97, 107, 118,
    130, 143, 157, 173, 190, 209, 230, 253, 279, 307,
    337, 371, 408, 449, 494, 544, 598, 658, 724, 796,
    876, 963, 1060, 1166, 1282, 1411, 1552, 1707, 1878, 2066,
    2272, 2499, 2749, 3024, 3327, 3660, 4026, 4428, 4871, 5358,
    5894, 6484, 7132, 7845, 8630, 9493, 10442, 11487, 12635, 13899,
    15289, 16818, 18500, 20350, 22385, 24623, 27086, 29794, 32767
};

/*************************************************************************
*       ADPCMEncoder - ADPCM encoder routine                             *
*************************************************************************
*       Input variables:                                                *
*               short sample - 16-bit signed speech sample              *
*               struct ADPCMstate *state - ADPCM structure              *
*       Return variables:                                               *
*               char - 8-bit number containing the 4-bit ADPCM code     *
*************************************************************************/
char ADPCMEncoder( short sample , struct ADPCMstate *state )
{
    int code;/* ADPCM output value */
    int diff;/* Difference between sample and the predicted
                                sample */
    int step;/* Quantizer step size */
    int predsample;/* Output of ADPCM predictor */
    int diffq;/* Dequantized predicted difference */
    int index;/* Index into step size table */

    /* Restore previous values of predicted sample and quantizer step
       size index
    */
    predsample = (int)(state->prevsample);
    index = state->previndex;
    step = StepSizeTable[index];


    /* Compute the difference between the acutal sample (sample) and the
       the predicted sample (predsample)
    */
    diff = sample - predsample;
    if(diff >= 0)
        code = 0;
```

```
else
{
    code = 8;
    diff = -diff;
}

/* Quantize the difference into the 4-bit ADPCM code using the
   the quantizer step size
*/
/* Inverse quantize the ADPCM code into a predicted difference
   using the quantizer step size
*/
diffq = step >> 3;
if( diff >= step )
{
    code |= 4;
    diff -= step;
    diffq += step;
}
step >>= 1;
if( diff >= step )
{
    code |= 2;
    diff -= step;
    diffq += step;
}
step >>= 1;
if( diff >= step )
{
    code |= 1;
    diffq += step;
}

/* Fixed predictor computes new predicted sample by adding the
   old predicted sample to predicted difference
*/
if( code & 8 )
    predsample -= diffq;
else
    predsample += diffq;

/* Check for overflow of the new predicted sample
*/
if( predsample > 32767 )
    predsample = 32767;
else if( predsample < -32767 )
    predsample = -32767;

/* Find new quantizer stepsize index by adding the old index
   to a table lookup using the ADPCM code
*/
index += IndexTable[code];

/* Check for overflow of the new quantizer step size index
*/
if( index < 0 )
    index = 0;
if( index > 88 )
    index = 88;

/* Save the predicted sample and quantizer step size index for
   next iteration
*/
state->prevsample = (short)predsample;
state->previndex = index;
```

```
    /* Return the new ADPCM code
    */
    return ( code & 0x0f );
}

/*******************************************************************************
*       ADPCMDecoder - ADPCM decoder routine                                   *
*******************************************************************************
*       Input variables:                                                       *
*               char code - 8-bit number containing the 4-bit ADPCM code       *
*               struct ADPCMstate *state - ADPCM structure                     *
*       Return variables:                                                      *
*               int - 16-bit signed speech sample                              *
*******************************************************************************/
int ADPCMDecoder(char code, struct ADPCMstate *state)
{
    int step;/* Quantizer step size */
    int predsample;/* Output of ADPCM predictor */
    int diffq;/* Dequantized predicted difference */
    int index;/* Index into step size table */

    /* Restore previous values of predicted sample and quantizer step
       size index
    */
    predsample = (int)(state->prevsample);
    index = state->previndex;

    /* Find quantizer step size from lookup table using index
    */
    step = StepSizeTable[index];

    /* Inverse quantize the ADPCM code into a difference using the
       quantizer step size
    */
    diffq = step >> 3;
    if( code & 4 )
        diffq += step;
    if( code & 2 )
        diffq += step >> 1;
    if( code & 1 )
        diffq += step >> 2;

    /* Add the difference to the predicted sample
    */
    if( code & 8 )
        predsample -= diffq;
    else
        predsample += diffq;

    /* Check for overflow of the new predicted sample
    */
    if( predsample > 32767 )
        predsample = 32767;
    else if( predsample < -32767 )
        predsample = -32767;

    /* Find new quantizer step size by adding the old index and a
       table lookup using the ADPCM code
    */
    index += IndexTable[code];

    /* Check for overflow of the new quantizer step size index
    */
    if( index < 0 )
        index = 0;
    if( index > 88 )
```

```
        index = 88;


    /* Save predicted sample and quantizer step size index for next
       iteration
    */
    state->prevsample = (short)predsample;
    state->previndex = index;

    /* Return the new speech sample */
    return( predsample );

}
```

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