Microsoft®

Multimedia

Standards Update

New Multimedia Data Types and Data Techniques

April 15, 1994
Revision: 3.0

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Overview

This standards update presents new and updated information for dealing with multimedia data under Microsoft Windows. This document is also available as part of the Multimedia Developer Registration Kit. The MDRK is used to register multimedia data and ids as well as new MCI command sets.. This document is the result of companies requesting and registering new data types. This document builds on the standard RIFF documentation that is contained in:

1. The Multimedia Development Kit (MDK) 1.0 Programmer's Reference
2. The Windows 3.1 Software Development Kit (SDK)'s Multimedia Programmer's Reference
3. The Multimedia Programmer's Reference book from Microsoft Press
4. Video for Windows 1.1 SDK Programmer's Guide

The RIFF file format is a standard published as a joint design document by IBM and Microsoft. This standards document is Multimedia Programming Interface and Data Specifications 1.0 published in August 1991. The first draft of this document was issued in November, 1990. This IBM/Microsoft document is available from the sources listed below.

This standards update assumes that the reader has read the concepts defined in these documents.

New RIFF file forms and chunks are defined in this document. The new RIFF forms and chunks defined here have been registered with Microsoft. If you want to register your own RIFF forms and chunks, fill out and return the Multimedia Developer Registration Kit included in this kit.

In addition, techniques for dealing with multimedia data in the system, such as clipboard data, are defined in this document.

Where to Look for Information

All constants and structures defined in this document are contained in MMREG.H, which is included in this kit.

Current versions of this document as well as other technical update and technical notes and sample code are available from the sources listed in the *Multimedia Document Overview*, included in this kit.

Versions of this Document

This document is continually being updated and expanded. Eventually the information presented in this document will be placed in the standard reference for the multimedia IDs standards from Microsoft, such as the Multimedia Programmer's Reference from MS-Press.

When refering to standards defined in this document, please refer to the data and version number printed on the cover page.

Please refer to the document *Microsoft Multimedia Document Overview*, which is included in this kit, for lists of other documents and sample code.

Version	Date	Who	Comment
1.0		Matt Saettler	Original release
2.1.0	June 22, 1993	Heidi Breslauer	Updated info
2.1.1	August 25, 1993	Heidi Breslauer	Updated info
2.1.3	September 5, 1993	Heidi Breslauer	Updates and corrections
3.0	April 5, 1994	Heidi Breslauer	Updates and reorganization

New RIFF Chunks

These new chunks have been defined for use in any RIFF form.

Display Chunk

Added: 05/01/92
Author: Microsoft

A DISP chunk contains easily rendered and displayable objects associated with an instance of a more complex object in a RIFF form (e.g. sound file, AVI movie).

A DISP chunk is defined as follows:

```
<DISP_ck>   →    DISP( <type> <data> )
```

<type> is a `DWORD` (32 bit unsigned quantity in Intel format) that identifies <data> as one of the standard Windows clipboard formats (`CF_METAFILE, CF_DIB, CF_TEXT`, etc.) as defined in *windows.h*.

The `DISP` chunk should be used as a direct child of the RIFF chunk so that any RIFF aware application can find it. There can be multiple `DISP` chunks with each containing different types of displayable data, but all representative of the same object. The `DISP` chunks should be stored in the file in order of preference (just as in the clipboard).

The `DISP` chunk is especially beneficial when representing OLE data within an application. For example, when pasting a wave file into Excel, the creating application can use the `DISP` chunk to associate an icon and a text description to represent the embedded wave file. This text should be short so that it can be easily displayed in menu bars and under icons.

Note: do not use a `CF_TEXT` for a description of the data. Bibliographic data chunks will be added to support the standard MARC (Machine Readable Cataloging) data.

JUNK (Filler) Chunk

Added: 05/01/92
Author: IBM, Microsoft
A JUNK chunk represents , filler or outdated information. It contains no relevant data; it is a space filler of arbitrary size. The JUNK chunk is defined as follows:

<JUNK chunk> ➠ JUNK(<filler>)
where **<filler>** contains random data.

PAD (Filler) Chunk

Added: 07/15/92
Author: Microsoft

A PAD chunk represents padding. It contains no relevant data; it is a space filler of arbitrary size. When duplicating the file, the copier should maintain the padding of the PAD chunk. Specifically, if the PAD chunk makes the next chunk align on a 2K boundary in the physical file, then this alignment should be preserved even if the size of the PAD chunk must change. The PAD chunk is defined as follows:

<PAD chunk> ➠ PAD(<filler>)
where **<filler>** contains random data.

New NFO list Chunks

For complete AVI file documentation, see the Multimedia Developer Reference, part of the Microsoft Windows SDK.

These chunks were added for Video for Windows 1.1, mid 1993:

ISMP	SMPTE time code of digitization start point expressed as a NULL terminated text string "HH:MM:SS.FF". If performing MCI capture in AVICAP, this chunk will be automatically set based on the MCI start time.
IDIT	"Digitization Time" Specifies the time and date that digitization commenced. The digitization time is contained in an ASCII string which contains exactly 26 characters and is in the format "Wed Jan 02 02:03:55 1990\n\0". The ctime(), asctime(), functions can be used to create strings in this format. This chunk is automatically added to the capture file based on the current system time at the moment capture is initiated.

New Forms

AVI

The RIFF AVI file format is defined in the Video for Windows SDK.

CPPO

Added: 12/16/92
Author: APPS Software International
4417 North Saddlebag Trail
Scottsdale, AZ 85251

Definition

CPPO RIFF Form Definition
(C) Copyright APPS Software International 1992
Revision 1.0

This document provides a new RIFF specification used to provide an object persistance/archival feature for Windows applications doped with C++. This RIFF form preserves not only the content of objects but also all the linkages between objects, providing a mechanism for loading and unloading an entire memory image.

```
<CPPO-form> --> RIFF ( 'CPPO'                    // RIFF form header
                    <object-list> )              // object list
```

```
<object-list> --> LIST ( 'obj' <object-ck> ... )
```

An <object-ck> can be one of the following:

1. an object referent (sub-chunk type 'objr').
2. an object instance (sub-chunk type 'obji').

'objr' chunks do not create a new instance of an object, but instead create a reference/pointer to the original instance of the specified object. Each object is numbered in the order that it appears in the file. The special object number zero represents a NULL pointer, thus the first object in the file is given the number 1.

```
<object-ck> --> objr ( <object-number:WORD> ) |    // object reference
                obji ( <object-instance> )         // new object instance
```

```
<object-instance> --> <class-descr> <member-list>
```

The <class-descr> reduces file size by specifying the class name only once. The first 'clsi' in the file is given the number 1.

```
<class-descr> --> clsr ( <class-number:WORD> ) |    // previously defined class
                  clsi ( <class-name:ZSTR> )        // new class definition
```

The remainder of the <object-instance> definition is its member list. A member list is a sequence of primitive data elements and/or object instances/references.

```
<member-list> --> LIST ( 'mbr' <member-ck> ... )
```

Each class is responsible for parsing its members from the RIFF file. It may choose to specify primitive data as a single 'byte' sub-chunk, or as a sequence of more specific chunks. Each non-primitive member must, however, be in the <object-ck> format.

```
<member-ck> --> <primitive-ck> |                // primitive data type
                <object-ck>                     // object definition

<primitive-ck> --> char ( <CHAR> ... ) |
                   byte ( <BYTE> ... ) |
                   int ( <INT> ... ) |
                   word ( <WORD> ... ) |
                   long ( <LONG> ... ) |
                   dwrd ( <DWORD> ... ) |
                   flt ( <FLOAT> ... ) |
                   dbl ( <DOUBLE> ... ) |
                   str ( <ZSTR> ... )
```

Example

CPPO RIFF Form Example
(C) Copyright APPS Software International 1992
Revision 1.0

This example stores an 'OrdCollect' object containing two 'String' objects and a NULL pointer.

```
RIFF ( 'CPPO'
        LIST ( 'INFO'
                INAM ( "Generic C++ Image"Z )
                ICOP ( "(C) Copyright APPS Software Int'l 1992"Z )
                ICRD ( "1992-12-10"Z )
            )
        LIST ( 'obj'
                obji (
                    clsi ( "String"Z )
                    LIST ( 'mbr'
                            str ( "This is the first"Z )
                        )
                    )
                obji (
                    clsr ( 1 )
                    LIST ( 'mbr'
                            str ( "This is the second"Z )
                        )
                    )
                obji (
                    clsi ( "OrdCollect"Z )
                    LIST ( 'mbr'
                            word ( 2 )
                            objr ( 1 )
                            objr ( 0 )
                            objr ( 2 )
                        )
                    )
            )
    )
```

ACON

Added: 4/13/93
Author: Microsoft

Windows NT Animated Cursor RIFF Files

For Windows NT, an animated cursor is stored in RIFF a file with a form type of 'ACON'. The subcunks of this form of RIFF file are the 'LIST', 'anih', 'rate', and 'seq ' chunks. There are two LIST chunks: the LIST chunk with type 'INFO' contains textual informative details about the animated cursor, the LIST chunk with a type of 'fram' contains 'icon' subchunks. The anih chunk describes the rest of the subchunks in the file. The 'rate' chunk tells how long each step of the animation is to be displayed on the screen. The 'seq ' chunk maps the animation steps into actual icon pictures stored in the .ani file. The 'icon' subchunks in the 'fram' LIST are the actual frames of the cursor animation.

The following is a RIFF grammar (as defined in the Microsoft Windows Multimedia Programmer's Reference) that describes the Windows NT animated cursors:

```
RIFF( 'ACON'
   [LIST( 'INFO' <info_data> )]
   [<DISP_ck>]
   anih( <ani_header> )
   [rate( <rate_info> )]
   ['seq '( <sequence_info> )]
   LIST( 'fram' icon( <icon_file> ) ... )
)
```

Where	Means:
info_data	Optional information subchunks as defined in the Microsoft Windows Multimedia Programmer's Reference.
DISP_ck	An optional Display Chunk as defined in the Multimedia Data Standards Update
ani_header	An animated cursor header (see ANIHEADER struct). The ANIHEADER structure describes the size of the animated cursor, and must come before the 'rate', 'seq ', or any 'icon' chunks.
rate_info	A list of JIF's (one for each step) that indicate the time that each step should remain on the screen. If this chunk is missing, then the default rate value specified in the ANIHEADER is used. A JIF is nothing more than a DWORD that holds the display time value in jiffies. One jiffy equals 1/60th of a second.
sequence_info	A list of indices into the 'fram' LIST of icons. Each index points to the icon that is to be displayed for that step of the animated cursor. The indices are 0 based. If this chunk is missing, then the icons are played in the order they occur in the file. To save space, it is common for the same icon to be referenced by different elements of the sequence list.9
icon_file	An embedded .CUR file as generated by the WindowsNT IMAGEDIT program. There are usually multiple icon subchunks in the file. Each icon represents a frame of the animated cursor. If present, the seq chunk determines the order in which the frames are displayed on the screen.

Structure Definitions:

```
typedef DWORD JIF;                  /* Number of jiffies that a frame
                                     * will remain on the screen
                                     */

typedef struct _ANIHEADER {     /* anih */
   DWORD cbSizeof;        /* Num. bytes in aniheader (incl. cbSizeof) */
   DWORD cFrames;         /* Number of unique icons in the ani. cursor*/
   DWORD cSteps;                  /* Number of blts before the animation cycles */
   DWORD cx, cy;                  /* reserved, must be 0 */
   DWORD cBitCount, cPlanes;      /* reserved, must be 0 */
   JIF   jifRate;    /* default rate if rate chunk not present */
   DWORD fl;                 /* flags, see AF_* */
} ANIHEADER, *PANIHEADER;
```

#define AF_ICON   0x0001L   /* Windows format icon/cursor animation */

Example

The following is a partial hex-dump of animated cursor file:
banana.ani:

```
00000000   52 49 46 46 78 2e 00 00 41 43 4F 4E 4c 49 53 54  RIFFx...ACONLIST
00000010   4a 00 00 00 49 4e 46 4f 49 4e 41 4d 0f 00 00 00  J...INFOINAM....
00000020   50 65 65 6c 69 6e 67 20 42 61 6e 61 6e 61 00 00  Peeling Banana..
00000030   49 41 52 54 26 00 00 00 4d 69 63 72 6f 73 6f 66  IART&...Microsof
00000040   74 20 43 6f 72 70 6f 72 61 74 69 6f 6e 2c 20 43  t Corporation, C
00000050   6f 70 79 72 69 67 68 74 20 31 39 39 33 00 61 6e  opyright 1993.an
00000060   69 68 24 00 00 00 24 00 00 00 0f 00 00 00 10 00  ih$...$.........
00000070   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00  ................
00000080   00 00 0f 00 00 00 03 00 00 00 72 61 74 65 40 00  ..........rate@.
00000090   00 00 0f 00 00 00 0f 00 00 00 0f 00 00 00 0f 00  ................
000000a0   00 00 0f 00 00 00 0f 00 00 00 0f 00 00 00 0f 00  ................
000000b0   00 00 0f 00 00 00 0f 00 00 00 0f 00 00 00 1e 00  ................
000000c0   00 00 14 00 00 00 14 00 00 00 32 00 00 00 1e 00  ..........2.....
000000d0   00 00 73 65 71 20 40 00 00 00 00 00 00 00 01 00  ..seq @.........
000000e0   00 00 02 00 00 00 03 00 00 00 04 00 00 00 05 00  ................
000000f0   00 00 06 00 00 00 07 00 00 00 08 00 00 00 09 00  ................
00000100   00 00 0a 00 00 00 0b 00 00 00 0c 00 00 00 0d 00  ................
00000110   00 00 0e 00 00 00 00 00 00 00 4c 49 53 54 5e 2d  ..........LIST^-
00000120   00 00 66 72 61 6d 69 63 6f 6e fe 02 00 00 00 00  ..framicon......
00000130   02 00 01 00 20 20 00 00 10 00 10 00 e8 02 00 00  ....  ..........
00000140   16 00 00 00 28 00 00 00 20 00 00 00 40 00 00 00  ....(... ...@...
00000150   01 00 04 00 00 00 00 00 80 02 00 00 00 00 00 00  ................
00000160   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00  ................
00000170   00 00 80 00 00 80 00 00 00 80 80 00 80 00 00 00  ................
00000180   80 00 80 00 80 80 00 00 80 80 80 00 c0 c0 c0 00  ................
00000190   00 00 ff 00 00 ff 00 00 00 ff ff 00 ff 00 00 00  ................
000001a0   ff 00 ff 00 ff ff 00 00 ff ff ff 00 00 00 00 00  ................
000001b0   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00  ................
000001c0   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00  ................
000001d0   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00  ................
000001e0   00 00 00 00 00 03 33 00 00 00 00 00 00 00 00 00  ......3.........
000001f0   00 00 00 00 03 3b ff 00 00 00 00 00 00 00 00 00  .....;..........
00000200   00 00 00 00 3b bf fb 30 00 00 00 00 00 00 00 00  ....;..0........
00000210   00 00 00 03 bb ff b3 00 00 00 00 00 00 00 00 00  ................
00000220   00 00 00 3b bf fb 30 00 00 00 00 00 00 00 00 00  ...;..0.........
00000230   00 00 03 bb ff b8 00 00 00 00 00 00 00 00 00 00  ................
00000240   00 00 03 bf fb 83 00 00 00 00 00 00 00 00 00 00  ................
00000250   00 00 3b bf bb 80 00 00 00 00 00 00 00 00 00 00  ..;.............
00000260   00 00 3b ff b8 30 00 00 00 00 00 00 00 00 00 00  ..;..0..........
00000270   00 03 bb fb b8 00 00 00 00 00 00 00 00 00 00 00  ................
00000280   00 03 bf fb b3 00 00 00 00 00 00 00 00 00 00 00  ................
00000290   00 03 bf bb 83 00 00 00 00 00 00 00 00 00 00 00  ................
000002a0   00 03 bf bb 80 00 00 00 00 00 00 00 00 00 00 00  ................
000002b0   00 03 bf bb 30 00 00 00 00 00 00 00 00 00 00 00  ....0...........
000002c0   00 03 bf bb 30 00 00 00 00 00 00 00 00 00 00 00  ....0...........
000002d0   00 03 bf bb 30 00 00 00 00 00 00 00 00 00 00 00  ....0...........
000002e0   00 03 bf bb 30 00 00 00 00 00 00 00 00 00 00 00  ....0...........
000002f0   00 03 bf bb 30 00 00 00 00 00 00 00 00 00 00 00  ....0...........
00000300   00 00 3f fb 30 00 00 00 00 00 00 00 00 00 00 00  ..?.0...........
00000310   00 00 3b fb 30 00 00 00 00 00 00 00 00 00 00 00  ..;.0...........
```

```
00000320   00 00 03 ff 30 00 00 00 00 00 00 00 00 00 00 00   ....0...........
00000330   00 00 03 bf 30 00 00 00 00 00 00 00 00 00 00 00   ....0...........
00000340   00 00 00 3f 80 00 00 00 00 00 00 00 00 00 00 00   ...?............
00000350   00 00 00 03 00 00 00 00 00 00 00 00 00 00 00 00   ................
00000360   00 00 00 00 33 00 00 00 00 00 00 00 00 00 00 00   ....3...........
00000370   00 00 00 00 03 30 00 00 00 00 00 00 00 00 00 00   .....0..........
00000380   00 00 00 00 00 30 00 00 00 00 00 00 00 00 00 00   .....0..........
00000390   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00   ................
000003a0   00 00 00 00 00 00 00 00 00 00 00 00 ff ff ff ff   ................
000003b0   ff ff ff ff ff ff ff ff ff ff e0 ff ff ff 80 ff   ................
000003c0   ff ff 00 ff ff fe 01 ff ff fc 03 ff ff f8 07 ff   ................
000003d0   ff f8 07 ff ff f0 0f ff ff f0 0f ff ff e0 1f ff   ................
000003e0   ff e0 1f ff ff e0 1f ff ff e0 3f ff ff e0 3f ff   .........?...?.
000003f0   ff e0 3f ff ff e0 3f ff ff e0 3f ff ff e0 3f ff   ..?...?...?...?.
00000400   ff f0 3f ff ff f0 3f ff ff f8 3f ff ff f8 3f ff   ..?...?...?...?.
00000410   ff fc 3f ff ff fe 3f ff ff ff 1f ff ff ff 8f ff   ..?...?.........
00000420   ff ff cf ff ff ff ff ff ff ff ff ff 69 63 6f 6e   ............icon
00000430   fe 02 00 00 00 00 02 00 01 00 20 20 00 00 10 00   ..........  ....
00000440   10 00 e8 02 00 00 16 00 00 00 28 00 00 00 20 00   ..........(... .
00000450   00 00 40 00 00 00 01 00 04 00 00 00 00 00 80 02   ..@.............
00000460   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00   ................
00000470   00 00 00 00 00 00 00 00 80 00 00 80 00 00 00 80   ................
00000480   80 00 80 00 00 00 80 00 80 00 80 80 00 00 80 80   ................
00000490   80 00 c0 c0 c0 00 00 00 ff 00 00 ff 00 00 00 ff   ................
000004a0   ff 00 ff 00 00 00 ff 00 ff 00 ff ff 00 00 ff ff   ................
000004b0   ff 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00   ................
000004c0   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00   ................
000004d0   00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00   ................
000004e0   00 00 00 00 00 00 00 00 00 00 00 03 33 00 00 00   ............3...
000004f0   00 00 00 00 00 00 00 00 00 00 00 03 3b ff 00 00 00   ...........;....
00000500   00 00 00 00 00 00 00 00 00 00 3b bf fb 30 00 00   ..........;..0..
00000510   00 00 00 00 00 00 00 00 00 03 bb ff b3 00 00 00   ................
00000520   00 00 00 00 00 00 00 00 00 3b bf fb 30 00 00 00   .........;..0...
00000530   00 00 00 00 00 00 00 00 03 bb ff b8 00 00 00 00   ................
00000540   00 00 00 00 00 00 00 00 03 bf fb 83 00 00 00 00   ................
...
(more embedded icon files)
...
00002e40   f8 00 00 7f fc 00 00 ff fe 00 21 ff ff e0 7f ff   ..........!.....
00002e50   ff e1 ff ff ff f3 ff ff ff ff ff ff ff ff ff ff   ................
00002e60   ff ff ff ff ff ff ff ff ff ff ff ff ff ff ff ff   ................
00002e70   ff ff ff ff ff ff ff ff ff ff ff ff ff ff ff ff   ................
```

New WAVE RIFF Chunks

Added: 05/01/92
Author: Microsoft, IBM

Most of the information in this section comes directly from the IBM/Microsoft RIFF standard document.

The WAVE form is defined as follows. Programs must expect (and ignore) any unknown chunks encountered, as with all RIFF forms. However, **<'fmt'-ck>** must always occur before **<wave-data>**, and both of these chunks are mandatory in a WAVE file.

```
<WAVE-form> ➠
        RIFF( 'WAVE'
                <'fmt'-ck>                      // Format
                [<fact-ck>]                     // Fact chunk
                [<cue-ck>]                      // Cue points
                [<playlist-ck>]                 // Playlist
                [<assoc-data-list>]             // Associated data list
                <wave-data>     )               // Wave data
```

The WAVE chunks are described in the following sections.

Fact Chunk

The **<fact-ck>** stores file dependent information about the contents of the WAVE file. This chunk is defined as follows:

```
<fact-ck>       →       fact( <dwSampleLength:DWORD> )
```

<**dwSampleLength**> represents the length of the data in samples. The <**nSamplesPerSec**> field from the wave format header is used in conjunction with the <**dwSampleLength**> field to determine the length of the data in seconds.

The `fact` chunk is required for all new `WAVE` formats. The chunk is not required for the standard WAVE_FORMAT_PCM files.

The `fact` chunk will be expanded to include any other information required by future WAVE formats. Added fields will appear following the **<dwSampleLength>** field. Applications can use the chunk size field to determine which fields are present.

Cue Points Chunk

The <**cue-ck**> cue-points chunk identifies a series of positions in the waveform data stream. The **<cue-ck>** is defined as follows:

```
<cue-ck> ➠          cue(    <dwCuePoints:DWORD>         // Count of cue points
                            <cue-point>... )            // Cue-point table

<cue-point> ➠       struct {
                            DWORD  dwName;
                            DWORD  dwPosition;
                            FOURCC fccChunk;
                            DWORD  dwChunkStart;
                            DWORD  dwBlockStart;
                            DWORD  dwSampleOffset;
                    }
```

The <**cue-point**> fields are as follows:

Field	Description
dwName	Specifies the cue point name. Each **<cue-point>** record must have a unique **dwName** field.
dwPosition	Specifies the sample position of the cue point. This is the sequential sample number within the play order. See “Playlist Chunk,” later in this document, for a discussion of the play order.
fccChunk	Specifies the name or chunk ID of the chunk containing the cue point.
dwChunkStart	Specifies the position of the start of the data chunk containing the cue point. This should be zero if there is only one chunk containing data (as is currently always the case).
dwBlockStart	Specifies the position of the start of the block containing the position. This is the byte offset from the start of the data section of the chunk, not the chunk's FOURCC.
dwSampleOffset	Specifies the sample offset of the cue point relative to the start of the block.

Examples of File Position Values

The following table describes the **<cue-point>** field values for a WAVE file containing a single ‘`data`’ chunk:

Cue Point Location	Field	Value
Within PCM data	**fccChunk**	FOURCC value ‘`data`’.
	dwChunkStart	Zero value.
	dwBlockStart	File position of the sample (nBlockAlign aligned bytes) relative to the start of the data section of the ‘`data`’ chunk (not the FOURCC).
	dwSampleOffset	Sample position of the cue point relative to the start of the ‘`data`’ chunk.
In all other ‘data’ chunks	**fccChunk**	FOURCC value ‘`data`’.
	dwChunkStart	Zero value.
	dwBlockStart	File position of the enclosing block relative to the start of the data section of the ‘`data`’ chunk (not the FOURCC). The software can begin the decompression at this point.
	dwSampleOffset	Sample position of the cue point relative to the start of the block.

Playlist Chunk

The **<playlist-ck>** playlist chunk specifies a play order for a series of cue points. The **<playlist-ck>** is defined as follows:

```
<playlist-ck> ➠          plst(
                                   <dwSegments:DWORD>  // Count of play
segments
                                   <play-segment>... )        // Play-segment table


<play-segment> ➠              struct {
                                        DWORD dwName;
                                        DWORD dwLength;
                                        DWORD dwLoops;
                                   }
```

The <**play-segment**> fields are as follows:

Field	Description
dwName	Specifies the cue point name. This value must match one of the names listed in the **<cue-ck>** cue-point table.
dwLength	Specifies the length of the section in samples.
dwLoops	Specifies the number of times to play the section.

Associated Data Chunk

The **<assoc-data-list>** associated data list provides the ability to attach information like labels to sections of the waveform data stream. The **<assoc-data-list>** is defined as follows:

```
<assoc-data-list> ➠                LIST(   'adtl'
                                                                 <labl-ck>
        // Label
                                                                 <note-ck>
        // Note
                                                                 <ltxt-ck> }
        // Text with data length

<labl-ck> ➠                                labl(    <dwName:DWORD>
                                                                 <data:ZSTR> )

<note-ck> ➠                                note(    <dwName:DWORD>
                                                                 <data:ZSTR> )

<ltxt-ck> ➠                                ltxt(    <dwName:DWORD>

        <dwSampleLength:DWORD>
                                                                 <dwPurpose:DWORD>
                                                                 <wCountry:WORD>
                                                                 <wLanguage:WORD>
                                                                 <wDialect:WORD>
                                                                 <wCodePage:WORD>
                                                                 <data:BYTE>... )
```

Label and Note Information

The ‘`labl`’ and ‘`note`’ chunks have similar fields. The ‘`labl`’ chunk contains a label, or title, to associate with a cue point. The ‘`note`’ chunk contains comment text for a cue point. The fields are as follows:

Field	Description
dwName	Specifies the cue point name. This value must match one of the names listed in the <**cue-ck**> cue-point table.
data	Specifies a NULL-terminated string containing a text label (for the ‘`labl`’ chunk) or comment text (for the ‘`note`’ chunk).

Text with Data Length Information

The “`ltxt`” chunk contains text that is associated with a data segment of specific length. The chunk fields are as follows:

Field	Description
dwName	Specifies the cue point name. This value must match one of the names listed in the <**cue-ck**> cue-point table.
dwSampleLength	Specifies the number of samples in the segment of waveform data.
dwPurpose	Specifies the type or purpose of the text. For example, <**dwPurpose**> can specify a FOURCC code like ‘scrp’ for script text or ‘`capt`’ for close-caption text.
wCountry	Specifies the country code for the text. See “Country Codes” for a current list of country codes.
wLanguage, **wDialect**	Specify the language and dialect codes for the text. See “Language and Dialect Codes” for a current list of language and dialect codes.
wCodePage	Specifies the code page for the text.

inst (Instrument) Chunk

Added: 12/29/92
Author: IBM
Defined for: WAVE form

The WAVE form is NEARLY the perfect file format for storing a sampled sound synthesizer's samples. Bits per sample, sample rate, number of channels, and complex looping can be specified with current WAVE subchunks, but a sample's pitch and its desired volume relative to other samples cannot. The optional instrument subchunk defined below fills in these needed parameters:

```
|<instrument-ck>| ➠ inst(
          <bUnshiftedNote:BYTE>
          <chFineTune:CHAR>
          <chGain:CHAR>
          <bLowNote:BYTE>
          <bHighNote:BYTE>
          <bLowVelocity:BYTE>
          <bHighVelocity:BYTE> )
```

bUnshiftedNote	the MIDI note number that corresponds to the unshifted pitch of the sample. Valid values range from 0 to 127.
chFineTune	the pitch shift adjustment in cents (or 100ths of a semitone) needed to hit bUnshiftedNote value exactly. chFineTune can be used to compensate for tuning errors in the sampling process. Valid values range from -50 to 50.
chGain	the suggested volume setting for the sample in decibels. A value of zero decibels suggests no change in the volume. A value of -6 decibels suggests reducing the amplitude of the sample by two.
bLowNote and bHigh Note	the suggested usable MIDI note number range of the sample. Valid values range from 0 to 127.
bLowVelocity and bHighVelocity	the suggested usable MIDI velocity range of the sample. Valid values range from 0 to 127.

smpl (Sample) Chunk

Added: 11/09/93
Author: Digidesign, Sonic Foundary, Turtle Beach
Defined for: WAVE form

The <sample-ck> sampled instrument chunk describes the minimum necessary information needed to allow a sampling keyboard to use a WAVE file as an instrument. Samplers which require more information can save their extended information in the sampler specific data section. The <sample-ck> is defined as follows:

```
|<sample-ck>| ➠ smpl(
          <dwManufacturer:DWORD>
          <dwProduct:DWORD>
          <dwSamplePeriod:DWORD>
          <dwMIDIUnityNote:DWORD>
          <dwMIDIPitchFraction:DWORD>
          <dwSMPTEFormat:DWORD>
          <dwSMPTEOffset:DWORD>
          <cSampleLoops:DWORD>
          <cbSamplerData:DWORD>
```

```
        <sample-loop(s)>
        <sampler-specific-data> )


<sample-loop>   struct
                        {
                                DWORD           dwIdentifier;
                                DWORD           dwType;
                                DWORD           dwStart;
                                DWORD           dwEnd;
                                DWORD           dwFraction;
                                DWORD           dwPlayCount;
                        }
```

The <sample-ck> chunk:

dwManufacturer	Specifies the MMA Manufacturer code for the intended target device. The high byte indicates the number of low order bytes (1 or 3) that are valid for the manufacturer code. For example, this value will be 0x01000013 for Digidesign (the MMA Manufacturer code is one byte, 0x13); whereas 0x03000041 identifies Microsoft (the MMA Manufacturer code is three bytes, 0x00 0x00 0x41). If the sample is not intended for a specific manufacturer, then this field should be set to zero.
dwProduct	Specifies the Product code of the intended target device for the dwManufacturer. If the sample is not intended for a specific manufacturer's product, then this field should be set to zero.
dwSamplePeriod	Specifies the period of one sample in nanoseconds (normally 1/nSamplesPerSec from the WAVEFORMAT structure for the RIFF WAVE file--however, this field allows fine tuning). For example, 44.1 kHz would be specified as 22675 (0x00005893).
dwMIDIUnityNote	Specifies the MIDI note which will replay the sample at original pitch. This value ranges from 0 to 127 (a value of 60 represents Middle C as defined by the MMA).
dwMIDIPitchFraction	Specifies the fraction of a semitone up from the specified dwMIDIUnityNote. A value of 0x80000000 is 1/2 semitone (50 cents); a value of 0x00000000 represents no fine tuning between semitones.
dwSMPTEFormat	Specifies the SMPTE time format used in the dwSMPTEOffset field. Possible values are (unrecognized formats should be ignored): 0 - specifies no SMPTE offset (dwSMPTEOffset should also be zero) 24 - specifies 24 frames per second 25 - specifies 25 frames per second 29 - specifies 30 frames per second with frame dropping ('30 drop') 30 - specifies 30 frames per second
dwSMPTEOffset	Specifies a time offset for the sample if it is to be syncronized or calibrated according to a start time other than 0. The format of this value is 0xhhmmssff. hh is a *signed* Hours value [-23..23]. mm is an unsigned Minutes value [0..59]. ss is unsigned Seconds value [0..59]. ff is an unsigned value [0..(<dwSMPTEFormat> - 1)].
cSampleLoops	Specifies the number (count) of <sample-loop> records that are contained in the <sample-ck> chunk. The <sample-loop> records are stored immediately following the cbSamplerData field.
cbSamplerData	Specifies the size in bytes of the optional <sampler-specific-data>. Sampler specific data is stored imediately following the <sample-loop> records. The cbSamplerData field will be zero if no extended sampler specific information is stored in the <sample-ck> chunk.

The <sample-loop> structure:

dwIdentifier	Identifies the unique 'name' of the loop. This field may correspond with a name stored in the <cue-ck> chunk. The name data is stored in the <assoc-data-list> chunk.
dwType	Specifies the loop type: 0 - Loop forward (normal) 1 - Alternating loop (forward/backward) 2 - Loop backward 3-31 - reserved for future standard types 32-? - sampler specific types (manufacturer defined)
dwStart	Specifies the startpoint of the loop in samples.
dwEnd	Specifies the endpoint of the loop in samples (this sample will also be played).
dwFraction	Allows fine-tuning for loop fractional areas between samples. Values range from 0x00000000 to 0xFFFFFFFF. A value of 0x80000000 represents 1/2 of a sample length.
dwPlayCount	Specifies the number of times to play the loop. A value of 0 specifies an infinite sustain loop.

New WAVE Types

The necessary type, structure and constant defintions are in *mmreg.h*.

All newly defined WAVE types must contain both a `fact` chunk and an extended wave format description within the `'fmt'` chunk. RIFF WAVE files of type WAVE_FORMAT_PCM need not have the extra chunk nor the extended wave format description.

Fact Chunk

This chunk stores file dependent information about the contents of the WAVE file. It currently specifies the length of the file in samples.

WAVEFORMATEX

The extended wave format structure is used to defined all non-PCM format wave data, and is described as follows in the include file *mmreg.h*:

```
/* general extended waveform format structure */
/* Use this for all NON PCM formats */
/* (information common to all formats) */
typedef struct waveformat_extended_tag {
   WORD    wFormatTag;        /* format type */
   WORD    nChannels;         /* number of channels (i.e. mono, stereo...) */
   DWORD   nSamplesPerSec;    /* sample rate */
   DWORD   nAvgBytesPerSec;   /* for buffer estimation */
   WORD    nBlockAlign;       /* block size of data */
   WORD    wBitsPerSample;    /* Number of bits per sample of mono data */
   WORD    cbSize;            /* The count in bytes of the extra size */} WAVEFORMATEX;
```

wFormatTag	Defines the type of WAVE file.
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 11025, 22050, or 44100. Other sample rates are allowed, but not encouraged. This rate is also used by the sample size entry in the fact chunk to determine the length in time of the data.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	The block alignment (in bytes) of the data in <**data-ck**>.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample per channel data. Each channel is assumed to have the same sample resolution. If this field is not needed, then it should be set to zero.
cbSize	The size in bytes of the extra information in the WAVE format header not including the size of the WAVEFORMATEX structure.. As an example, in the IMA ADPCM format cbSize is calculated as sizeof(IMAADPCMWAVEFORMAT) - sizeof(WAVEFORMATEX) which yeilds two.

Defined wFormatTags

Expr1	WAVE form Registration No - Hex	Expr2
#define WAVE_FORMAT_G723_ADPCM	0x0014	/* Antex Electronics Corporation */
#define WAVE_FORMAT_ANTEX_ADPCME	0x0033	/* Antex Electronics Corporation */
#define WAVE_FORMAT_G721_ADPCM	0x0040	/* Antex Electronics Corporation */
#define WAVE_FORMAT_APTX	0x0025	/* Audio Processing Technology */
#define WAVE_FORMAT_AUDIOFILE_AF36	0x0024	/* Audiofile, Inc. */
#define WAVE_FORMAT_AUDIOFILE_AF10	0x0026	/* Audiofile, Inc. */
#define WAVE_FORMAT_CONTROL_RES_VQLPC	0x0034	/* Control Resources Limited */
#define WAVE_FORMAT_CONTROL_RES_CR10	0x0037	/* Control Resources Limited */
#define WAVE_FORMAT_CREATIVE_ADPCM	0x0200	/* Creative Labs, Inc */
#define WAVE_FORMAT_DOLBY_AC2	0x0030	/* Dolby Laboratories */
#define WAVE_FORMAT_DSPGROUP_TRUESPEECH	0x0022	/* DSP Group, Inc */
#define WAVE_FORMAT_DIGISTD	0x0015	/* DSP Solutions, Inc. */
#define WAVE_FORMAT_DIGIFIX	0x0016	/* DSP Solutions, Inc. */
#define WAVE_FORMAT_DIGIREAL	0x0035	/* DSP Solutions, Inc. */
#define WAVE_FORMAT_DIGIADPCM	0x0036	/* DSP Solutions, Inc. */
#define WAVE_FORMAT_ECHOSC1	0x0023	/* Echo Speech Corporation */
#define WAVE_FORMAT_FM_TOWNS_SND	0x0300	/* Fujitsu Corp. */
#define WAVE_FORMAT_IBM_CVSD	0x0005	/* IBM Corporation */
#define WAVE_FORMAT_OLIGSM	0x1000	/* Ing C. Olivetti & C., S.p.A. */
#define WAVE_FORMAT_OLIADPCM	0x1001	/* Ing C. Olivetti & C., S.p.A. */
#define WAVE_FORMAT_OLICELP	0x1002	/* Ing C. Olivetti & C., S.p.A. */
#define WAVE_FORMAT_OLISBC	0x1003	/* Ing C. Olivetti & C., S.p.A. */
#define WAVE_FORMAT_OLIOPR	0x1004	/* Ing C. Olivetti & C., S.p.A. */
#define WAVE_FORMAT_IMA_ADPCM	(WAVE_FORM_DVI_ADPCM)	/* Intel Corporation */
#define WAVE_FORMAT_DVI_ADPCM	0x0011	/* Intel Corporation */
#define WAVE_FORMAT_UNKNOWN	0x0000	/* Microsoft Corporation */
#define WAVE_FORMAT_PCM	0x0001	/* Microsoft Corporation */
#define WAVE_FORMAT_ADPCM	0x0002	/* Microsoft Corporation */
#define WAVE_FORMAT_ALAW	0x0006	/* Microsoft Corporation */
#define WAVE_FORMAT_MULAW	0x0007	/* Microsoft Corporation */
#define WAVE_FORMAT_GSM610	0x0031	/* Microsoft Corporation */
#define WAVE_FORMAT_MPEG	0x0050	/* Microsoft Corporation */
#define WAVE_FORMAT_NMS_VBXADPCM	0x0038	/* Natural MicroSystems */
#define WAVE_FORMAT_OKI_ADPCM	0x0010	/* OKI */
#define WAVE_FORMAT_SIERRA_ADPCM	0x0013	/* Sierra Semiconductor Corp */
#define WAVE_FORMAT_SONARC	0x0021	/* Speech Compression */
#define WAVE_FORMAT_MEDIASPACE_ADPCM	0x0012	/* Videologic */
#define WAVE_FORMAT_YAMAHA_ADPCM	0x0020	/* Yamaha Corporation of America */

Unknown Wave Type

Added: 05/01/92
Author: Microsoft

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

Changed as of September 5, 1993: This wave format will not be defined. For development purposes, DO NOT USE 0x0000. Instead, USE 0xffff until an ID has been obtained.

#define WAVE_FORMAT_UNKNOWN (0x0000)

wFormatTag	This must be set to WAVE_FORMAT_UNKNOWN.
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

Microsoft ADPCM

Added 05/01/92
Author: Microsoft

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_ADPCM (0x0002)

```
typedef struct adpcmcoef_tag {
      int         iCoef1;
      int         iCoef2;
} ADPCMCOEFSET;

typedef struct adpcmwaveformat_tag {
      WAVEFORMATEX      wfxx;
      WORD              wSamplesPerBlock;
      WORD              wNumCoef;
      ADPCMCOEFSET      aCoeff[wNumCoef];
} ADPCMWAVEFORMAT;
```

wFormatTag	This must be set to WAVE_FORMAT_ADPCM.	
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.	
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 11025, 22050, or 44100. Other sample rates are allowed, but not encouraged.	
nAvgBytesPerSec	Average data rate. ((nSamplesperSec / nSamplesPerBlock) * nBlockAlign).	
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.	
nBlockAlign	The block alignment (in bytes) of the data in <data-ck>.	
	nSamplesPerSec x Channels	nBlockAlign
	8k	256
	11k	256
	22k	512
	44k	1024
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.	
wBitsPerSample	This is the number of bits per sample of ADPCM. Currently only 4 bits per sample is defined. Other values are reserved.	
cbSize	The size in bytes of the extended information after the WAVEFORMATEX structure.	
	For the standard WAVE_FORMAT_ADPCM using the standard seven coefficient pairs, this is 32. If extra coefficients are added, then this value will increase.	
nSamplesPerBlock	Count of number of samples per block.	
	`(((nBlockAlign - (7 * nChannels)) * 8) / (wBitsPerSample * nChannels)) + 2.`	
nNumCoef	Count of the number of coefficient sets defined in aCoef.	

aCoeff	These are the coefficients used by the wave to play. They may be interpreted as fixed point 8.8 signed values. Currently there are 7 preset coefficient sets. They must appear in the following order.		
	Coef Set	Coef1	Coef2
	0	256	0
	1	512	-256
	2	0	0
	3	192	64
	4	240	0
	5	460	-208
	6	392	-232
	Note that if even only 1 coefficient set was used to encode the file then all coefficient sets are still included. More coefficients may be added by the encoding software, but the first 7 must always be the same.		

Note: 8.8 signed values can be divided by 256 to obtain the integer portion of the value.

Block

The block has three parts, the header, data, and padding. The three together are <**nBlockAlign**> bytes.

```
typedef struct adpcmblockheader_tag {
      BYTE              bPredictor[nChannels];
      int               iDelta[nChannels];
      int               iSamp1[nChannels];
      int               iSamp2[nChannels];
} ADPCMBLOCKHEADER;
```

Field	**Description**
bPredictor	Index into the aCoef array to define the predictor used to encode this block.
iDelta	Initial Delta value to use.
iSamp1	The second sample value of the block. When decoding this will be used as the previous sample to start decoding with.
iSamp2	The first sample value of the block. When decoding this will be used as the previous' previous sample to start decoding with.

Data

The data is a bit string parsed in groups of (wBitsPerSample * nChannels).

For the case of Mono Voice ADPCM (wBitsPerSample = 4, nChannels = 1) we have:
`<Byte1> <Byte2>...<ByteN> ...<Byte((nSamplesPerBlock-2)/2)>`
where <ByteN> has <High Order Bit ... Low OrderBit> or < (Sample 2N + 2) (Sample 2N + 3)>

```
      <ByteN> =  ((4 bit error delta for sample (2 * N) + 2) <<
      4) | (4 bit error delta for sample (2 * N) + 3)
```

For the case of Stereo Voice ADPCM (wBitsPerSample = 4, nChannels = 2) we have:
`<Byte1> <Byte2>...<ByteN> ...<Byte(nSamplesPerBlock-2)>`
where <ByteN> has <High Order Bit ... Low OrderBit> or
< (Left Channel of Sample N + 2) (Right Channel of Sample N + 2)>

```
      <ByteN> =  ((4 bit error delta for left channel of sample
      N + 2) << 4)  |  (4 bit error delta for right channel of
      sample N + 2)
```

Padding

Bit Padding is used to round off the block to an exact byte length.
The size of the padding (in bits):

```
((nBlockAlign - (7 * nChannels)) * 8)    -
       (((nSamplesPerBlock - 2) * nChannels) * wBitsPerSample)
```

The padding does not store any data and should be made zero.

ADPCM Algorithm

Each channel of the ADPCM file can be encoded/decoded independently. However this should not destroy phase and amplitude information since each channel will track the original. Since the channels are encoded/decoded independently, this document is written as if only one channel is being decoded. Since the channels are interleaved, multiple channels may be encoded/decoded in parallel using independent local storage and temporaries.

Note that the process for encoding/decoding one block is independent from the process for the next block. Therefore the process is described for one block only, and may be repeated for other blocks. While some optimizations may relate the process for one block to another, in theory they are still independent.

Note that in the description below the number designation appended to iSamp (i.e. iSamp1 and iSamp2) refers to the placement of the sample in relation to the current one being decoded. Thus when you are decoding sample N, iSamp1 would be sample N - 1 and iSamp2 would be sample N - 2. Coef1 is the coefficient for iSamp1 and Coef2 is the coefficient for iSamp2. This numbering is identical to that used in the block and format descriptions above.

A sample application will be provided to convert a RIFF waveform file to and from ADPCM and PCM formats.

Decoding

First the predictor coefficients are determined by using the bPredictor field of block header. This value is an index into the aCoef array in the file header.

```
            bPredictor = GETBYTE
```

The initial iDelta is also taken from the block header.

```
            iDelta = GETWORD
```

Then the first two samples are taken from block header. (They are stored as 16 bit PCM data as iSamp1 and iSamp2. iSamp2 is the first sample of the block, iSamp1 is the second sample.)

```
            iSamp1= GETINT
            iSamp2 = GETINT
```

After taking this initial data from the block header, the process of decoding the rest of the block may begin. It can be done in the following manner:

```
While there are more samples in the block to decode:
Predict the next sample from the previous two samples.
      lPredSamp =  ((iSamp1 * iCoef1) + (iSamp2 *iCoef2)) /
  FIXED_POINT_COEF_BASE
Get the 4 bit signed error delta.
      (iErrorDelta = GETNIBBLE)
```

```
Add the 'error in prediction' to the predicted next sample and prevent over/underflow errors.
        (lNewSamp =  lPredSample + (iDelta * iErrorDelta)
        if lNewSample too large, make it the maximum allowable size.
        if lNewSample too small, make it the minimum allowable size.
Output the new sample.
        OUTPUT( lNewSamp )
Adjust the quantization step size used to calculate the 'error in prediction'.
        iDelta = iDelta * AdaptionTable[ iErrorDelta] / FIXED_POINT_ADAPTION_BASE
        if iDelta too small, make it the minimum allowable size.
Update the record of previous samples.
        iSamp2 = iSamp1;
        iSamp1 = lNewSample.
```

Encoding

For each block, the encoding process can be done through the following steps. (for each channel)

Determine the predictor to use for the block.
Determine the initial iDelta for the block.
Write out the block header.
Encode and write out the data.

The predictor to use for each block can be determined in many ways.

1. A static predictor for all files.
2. The block can be encoded with each possible predictor. Then the predictor that gave the least error can be chosen. The least error can be determined from:
    1. Sum of squares of differences. (from compressed/decompressed to original data)
    2. The least average absolute difference.
    3. The least average iDelta
3. The predictor that has the smallest initial iDelta can be chosen. (This is an approximation of method 2.3)
4. Statistics from either the previous or current block. (e.g. a linear combination of the first 5 samples of a block that corresponds to the average predicted error.)

The starting iDelta for each block can also be determined in a couple of ways.

1. One way is to always start off with the same initial iDelta.
2. Another way is to use the iDelta from the end of the previous block. (Note that for the first block an initial value must then be chosen.)
3. The initial iDelta may also be determined from the first few samples of the block. (iDelta generally fluctuates around the value that makes the absolute value of the encoded output about half maximum absolute value of the encoded output. (for 4 bit error deltas the maximum absolute value is 8. This means the initial iDelta should be set so that the first output is around 4.)
4. Finally the initial iDelta for this block may be determined from the last few samples of the last block. (Note that for the first block an initial value must then be chosen.)

Note that different choices for predictor and initial iDelta will result in different audio quality.

Once the predictor and starting quantization values are chosen, the block header may be written out.
First the choice of predictor is written out. (For each channel.)
Then the initial iDelta (quantization scale) is written out. (For each channel.)
Then the 16 bit PCM value of the second sample is written out. (iSamp1) (For each channel.)
Finally the 16 bit PCM value of the first sample is written out. (iSamp2) (For each channel.)

Then the rest of the block may be encoded. (Note that the first encoded value will be for the 3rd sample in the block since the first two are contained in the header.)

```
While there are more samples in the block to decode:
    Predict the next sample from the previous two samples.
        lPredSamp =  ((iSamp1 * iCoef1) + (iSamp2 *iCoef2))
                            / FIXED_POINT_COEF_BASE
    The 4 bit signed error delta is produced and overflow/underflow is prevented..
        iErrorDelta =  (Sample(n) - lPredSamp) / iDelta
        if iErrorDelta is too large, make it the maximum allowable size.
        if iErrorDelta is too small, make it the minimum allowable size.
    Then the nibble iErrorDelta is written out.
        PutNIBBLE( iErrorDelta )
     Add the 'error in prediction' to the predicted next sample and prevent over/underflow
        errors.
        (lNewSamp =  lPredSample + (iDelta * iErrorDelta)
        if lNewSample too large, make it the maximum allowable size.
        if lNewSample too small, make it the minimum allowable size.
    Adjust the quantization step size used to calculate the 'error in prediction'.
        iDelta = iDelta * AdaptionTable[ iErrorDelta] /
            FIXED_POINT_ADAPTION_BASE
        if iDelta too small, make it the minimum allowable size.
    Update the record of previous samples.
        iSamp2 = iSamp1;
        iSamp1 = lNewSample.
```

Sample C Code

Sample C Code is contained in the file msadpcm.c, which is available with this document in electronic form and separately. See the Overview section for how to obtain this sample code.

CVSD Wave Type

Added 07/21/92
Author: DSP Solutions, formerly Digispeech

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_IBM_CVSD (0x0005)

wFormatTag	This must be set to WAVE_FORMAT_IBM_CVSD
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo...
nSamplesPerSec	Frequency the source was sampled at. See chart below.
nAvgBytesPerSec	Average data rate. See chart below. (One of 1800, 2400, 3000, 3600, 4200, or 4800)
	Playback software can estimate the buffer size using the `<nAvgBytesPerSec>` value.
nBlockAlign	Set to 2048 to provide efficient caching of file from CD-ROM.
	Playback software needs to process a multiple of `<nBlockAlign>` bytes of data at a time, so that the value of `<nBlockAlign>` can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data. This is always 1 for CVSD.
cbSize	The size in bytes of the rest of the wave format header. This is zero for CVSD.

The Digispeech CVSD compression format is compatible with the IBM PS/2 Speech Adapter, which uses a Motorola MC3418 for CVSD modulation. The Motorola chip uses only one algorithm which can work at variable sampling clock rates. The CVSD algorithm compresses each input audio sample to 1 bit. An acceptable quality of sound is achieved using high sampling rates. The Digispeech DS201 adapter supports six CVSD sampling frequencies, which are being used by most software using the IBM PS/2 Speech Adapter:

Sample Rate	Bytes/Second
14,400Hz	1800 Bytes
19,200Hz	2400 Bytes
24,000Hz	3000 Bytes
28,800Hz	3600 Bytes
33,600Hz	4200 Bytes
38,400Hz	4800 Bytes

The CVSD format is a compression scheme which has been used by IBM and is supported by the IBM PS/2 Speech Adapter card. Digispeech also has a card that uses this compression scheme. It is not Digispeech's policy to disclose any of these algorithms to any third party vendor.

CCITT Standard Companded Wave Types

Added: 05/22/92
Author: Microsoft, DSP Solutions formerly Digispeech, Vocaltec, Artisoft

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

```
#define         WAVE_FORMAT_ALAW        (0x0006)
#define         WAVE_FORMAT_MULAW       (0x0007)
```

wFormatTag	This must be set to one of WAVE_FORMAT_ALAW, WAVE_FORMAT_MULAW
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo...
nSamplesPerSec	Frequency of the wave file. (8000, 11025, 22050, 44100).
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Size of the blocks in bytes.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data. (This is 8 for all the companded formats.)
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be zero.

See the CCITT G.711 specification for details of the data format.
This is a CCITT (International Telegraph and Telephone Consultative Committee) specification. Their address is:

Palais des Nations
CH-1211 Geneva 10, Switzerland
Phone: 22 7305111

OKI ADPCM Wave Types

Added: 05/22/92
Author: DigiSpeech, Vocaltec, Wang

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_OKI_ADPCM (0x0010)

```
typedef struct oki_adpcmwaveformat_tag {
        WAVEFORMATEX        wfx;
        WORD  wPole;
} OKIADPCMWAVEFORMAT;
```

wFormatTag	This must be set to WAVE_FORMAT_OKI_ADPCM		
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.		
nSamplesPerSec	Frequency the sample rate of the wave file. (8000, 11025, 22050, 44100).		
nAvgBytesPerSec	Average data rate.		
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.		
nBlockAlign	This is dependent upon the number of bits per sample.		
	wBitsPerSample	nChannels	nBlockAlign
	3	1	3
	3	2	6
	4	1	1
	4	2	1
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.		
wBitsPerSample	This is the number of bits per sample of data. (OKI can be 3 or 4)		
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 2.		
wPole	High frequency emphasis value		

This format is created and read by the OKI APDCM chip set. This chip set is used by a number of card manufacturers.

IMA ADPCM Wave Type

The IMA ADPCM and the DVI ADPCM are identical. Please see the following section on the DVI ADPCM Wave Type for a full description.

```
#define      WAVE_FORMAT_IMA_ADPCM                (0x0011)
```

DVI ADPCM Wave Type

Added: 12/16/92
Author: Intel
Please note that DVI ADPCM Wave Type is Identical to IMA ADPCM Wave Type.

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

```
#define      WAVE_FORMAT_DVI_ADPCM                (0x0011)
```

```
typedef struct dvi_adpcmwaveformat_tag {
        WAVEFORMATEX     wfx;
        WORD             wSamplesPerBlock;
} DVIADPCMWAVEFORMAT;
```

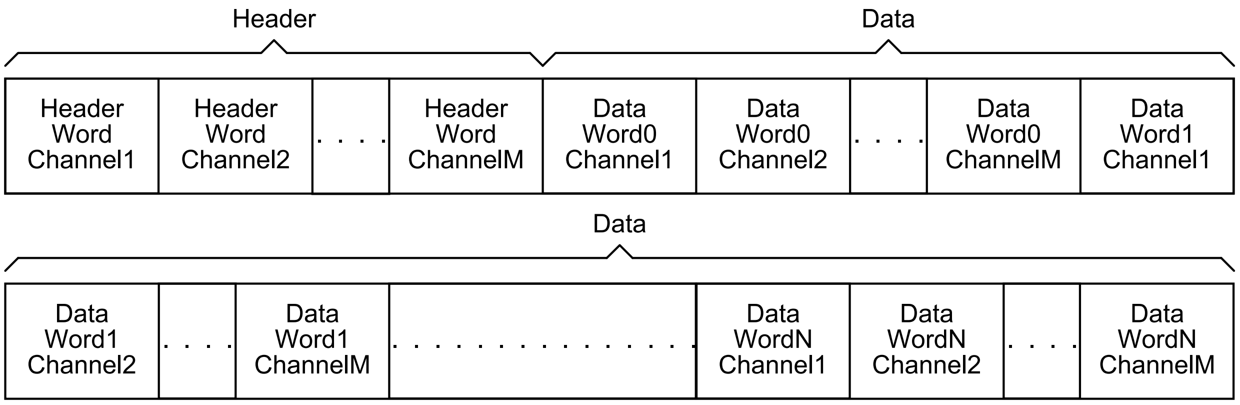
wFormatTag	This must be set to WAVE_FORMAT_DVI_ADPCM.	
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo...	
nSamplesPerSec	Sample rate of the WAVE file. This should be 8000, 11025, 22050 or 44100. Other sample rates are allowed.	
nAvgBytesPerSec	Total average data rate.	
	Playback software can estimate the buffer size for a selected amount of time by using the <**nAvgBytesPerSec**> value.	
nBlockAlign	This is dependent upon the number of bits per sample.	
	wBitsPerSample	nBlockAlign
	3	((N * 3) + 1) * 4 * nChannels
	4	(N + 1) * 4 * nChannels
		Where N = 0, 1, 2, 3 . . .
	The recommended block size for coding is 256 * <**nChannels**> bytes* min(1, (<**nSamplesPerSecond**>/ 11 kHz)) Smaller values cause the block header to become a more significant storage overhead. But, it is up to the implementation of the coding portion of the algorithm to decide the optimal value for <**nBlockAlign**> within the given constraints (see above). The decoding portion of the algorithm must be able to handle any valid block size. Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so the value of <**nBlockAlign**> can be used for allocating buffers.	
wBitsPerSample	This is the number of bits per sample of data. DVI ADPCM supports 3 or 4 bits per sample.	
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 2.	
wSamplesPerBlock	Count of the number of samples per channel per Block.	

	$\text{wSamplesPerBlock} = \frac{((\text{nBlockAlign} - (4 * \text{nChannels})) * 8}{\text{wBitsPerSample} * \text{nChannels}} + 1$

Block

The block is defined to be <**nBlockAlign**> bytes in length. For DVI ADPCM this must be a multiple of 4 bytes since all information in the block is divided on 32 bit word boundaries.

The block has two parts, the header and the data. The two together are <**nBlockAlign**> bytes in length. The following diagram shows the Header and Data parts of one block.



Where:

$$M = <\text{nChannels}>$$
$$N = \frac{< \text{nBlockAlign} >}{4* < \text{nChannels} >} - 1$$

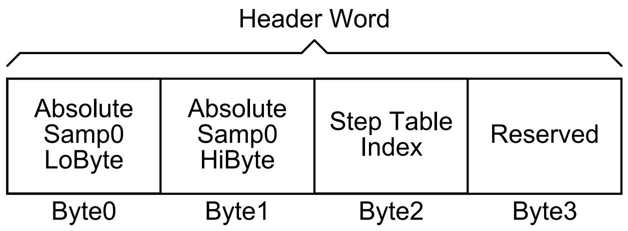
Header

This is a C structure that defines the DVI ADPCM block header.

```
typedef struct dvi_adpcmblockheader_tag {
    int         iSamp0;
    BYTE        bStepTableIndex;
    BYTE        bReserved;
} DVI_ADPCMBLOCKHEADER;
```

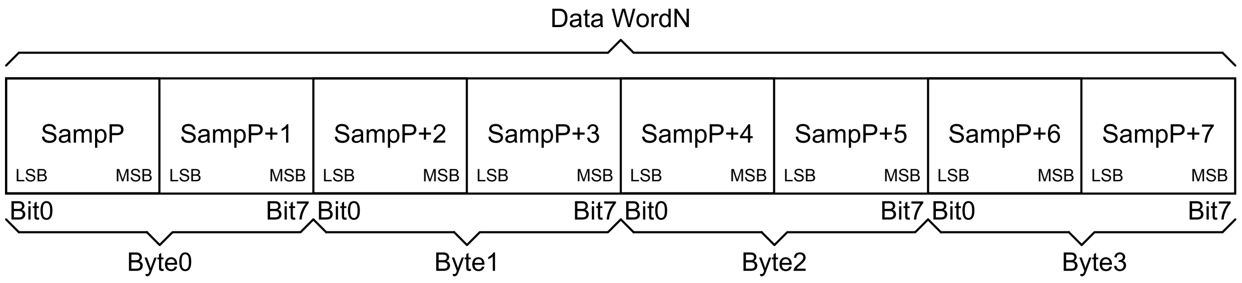
Field	Description
iSamp0	The first sample value of the block. When decoding, this will be used as the previous sample to start decoding with.
bStepTableIndex	The current index into the step table array. (0 - 88)
bReserved	This byte is reserved for future use.

A block contains an array of <**nChannels**> header structures as defined above. This diagram gives a byte level description of the contents of each header word.



Data

The data words are interpreted differently depending on the number of bits per sample selected.
For 4 bit DVI ADPCM (where <**wBitsPerSample**> is equal to four) each data word contains eight sample codes as shown in the following diagram.

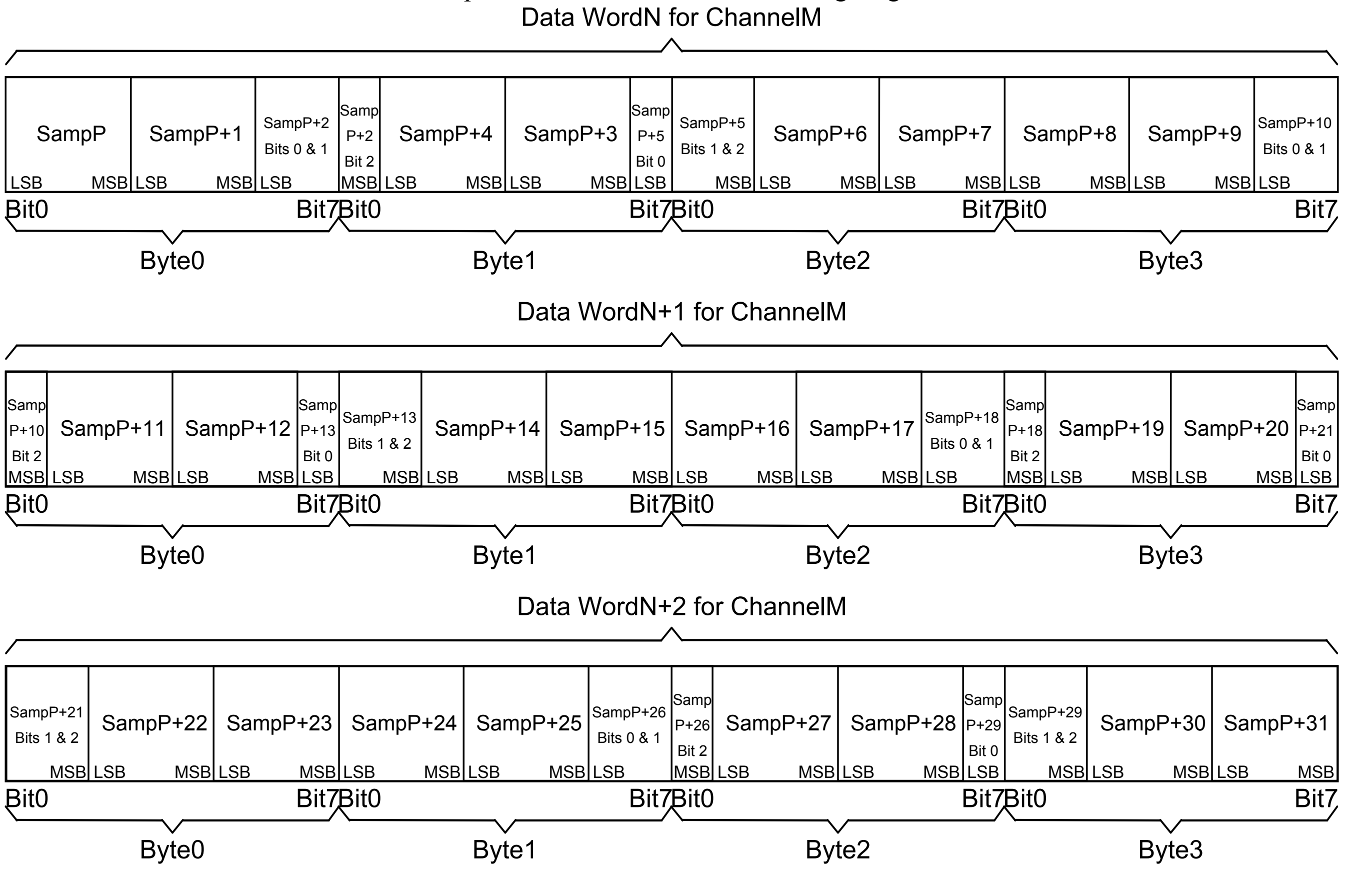


Where:
N = A data word for a given channel, in the range of 0 to
    <nBlockAlign> / (4 * <nChannels>) - <nChannels> - 1
P = (N * 8) + 1

Sample 0 is always included in the block header for the channel.
Each Sample is 4 bits in length. Each block contains a total of <**wSamplesPerBlock**> samples for each channel.

For 3 bit DVI ADPCM (where <**wBitsPerSample**> is equal to three) each data word contains 10.667 sample codes. It takes three words to hold an integral number of sample codes at 3 bits per code. So for

3 bit DVI ADPCM, the number of data words is required to be a multiple of three words (12 bytes). These three words contain 32 sample codes as shown in the following diagram.



Where:

`M` = One of the channels, in the range of 1 to <nChannels>
`N` = The first data word in a group of three data words for channelM, in the
range of 0 to <nBlockAlign> / (4 * <nChannels>) - <nChannels> - 1
`P = ( ( N / 3 ) * 32 ) + 1`

Sample 0 is always included in the block header for the channel.

Each Sample is 3 bits in length. Each block contains a total of <**wSamplesPerBlock**> samples for each channel.

DVI ADPCM Algorithm

Each channel of the DVI ADPCM file can be encoded/decoded independently. Since the channels are encoded/decoded independently, this document is written as if only one channel is being decoded. Since the channels are interleaved, multiple channels may be encoded/decoded in parallel using independent local storage and temporaries.

Note that the process for encoding/decoding one block is independent from the process for the next block. Therefore the process is described for one block only, and may be repeated for other blocks.

The processes for encoding and decoding is discussed below.

Tables

The DVI ADPCM algorithm relies on two tables to encode and decode audio samples. These are the step table and the index table. The contents of these tables are fixed for this algorithm. The 3 and 4 bit versions of the DVI ADPCM algorithm use the same step table, which is:

```
const int   StepTab[ 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 }
```

But, the index table is different for the different bit rates. For the 4 bit DVI ADPCM the contents of index table is:

```
const int   IndexTab[ 16 ] = { -1, -1, -1, -1, 2, 4, 6, 8,
                               -1, -1, -1, -1, 2, 4, 6, 8 };
```

For 3 bit DVI ADPCM the contents of the index table is:

```
const int   IndexTab[ 8 ] = { -1, -1, 1, 2,
                              -1, -1, 1, 2 };
```

Decoding

This section describes the algorithm used for decoding the 4 bit DVI ADPCM. This procedure must be followed for each block for each channel.

```
Get the first sample, Samp0, from the block header
Set the initial step table index, Index, from the block header
Output the first sample, Samp0
Set the previous Sample value:
    SampX-1 = Samp0
While there are still samples to decode
    Get the next sample code, SampX Code
    Calculate the new sample:
        Calculate the difference:
            Diff = 0
            if ( SampX Code & 4 )
                Diff = Diff + StepTab[ Index ]
            if ( SampX Code & 2 )
                Diff = Diff + ( StepTab[ Index ] >> 1 )
            if ( SampX Code & 1 )
                Diff = Diff + ( StepTab[ Index ] >> 2 )
            Diff = Diff + ( StepTab[ Index ] >> 3 )
        Check the sign bit:
            if ( SampX Code & 8 )
                Diff = -Diff
        SampX = SampX-1 + Diff
        Check for overflow and underflow errors:
                if **SampX** too large, make it the maximum allowable size (32767)
                if **SampX** too small, make it the minimum allowable size (-32768)
        Output the new sample, **SampX**
        Adjust the step table index:
                **Index** = **Index** + **IndexTab**[ **SampX Code** ]
        Check for step table index overflow and underflow:
                if **Index** too large, make it the maximum allowable size (88)
                if **Index** too small, make it the minimum allowable size (0)
        Save the previous Sample value:
                **SampX-1** = **SampX**
```

This section describes the algorithm used for decoding the 3 bit DVI ADPCM. This procedure must be followed for each block for each channel.

```
Get the first sample, **Samp0**, from the block header
Set the initial step table index, **Index**, from the block header
Output the first sample, **Samp0**
Set the previous Sample value:
        **SampX-1** = **Samp0**
While there are still samples to decode
        Get the next sample code, **SampX Code**
        Calculate the new sample:
                Calculate the difference:
                        **Diff** = 0
                        if ( **SampX Code** & 2 )
                                **Diff** = **Diff** + **StepTab**[ **Index** ]
                        if ( **SampX Code** & 1 )
                                **Diff** = **Diff** + ( **StepTab**[ **Index** ] >> 1 )
                        **Diff** = **Diff** + ( **StepTab**[ **Index** ] >> 2 )
                Check the sign bit:
                        if ( **SampX Code** & 4 )
                                **Diff** = **-Diff**
                **SampX** = **SampX-1** + **Diff**
        Check for overflow and underflow errors:
                if **SampX** too large, make it the maximum allowable size (32767)
                if **SampX** too small, make it the minimum allowable size (-32768)
        Output the new sample, **SampX**
        Adjust the step table index:
                **Index** = **Index** + **IndexTab**[ **SampX Code** ]
        Check for step table index overflow and underflow:
                if **Index** too large, make it the maximum allowable size (88)
                if **Index** too small, make it the minimum allowable size (0)
        Save the previous Sample value:
                **SampX-1** = **SampX**
```

Encoding

This section describes the algorithm used for encoding the 4 bit DVI ADPCM. This procedure must be followed for each block for each channel.

```
For the first block only, clear the initial step table index:
        **Index** = 0

Get the first sample, **Samp0**
```

```
Create the block header:
    Write the first sample, Samp0, to the header
    Write the initial step table index, Index, to the header
Set the previously predicted sample value:
    PredSamp = Samp0
While there are still samples to encode, and we're not at the end of the block
    Get the next sample to encode, SampX
    Calculate the new sample code:
        Diff = SampX - PredSamp
        Set the sign bit:
            if ( Diff < 0 )
                SampX Code = 8
                Diff = -Diff
            else
                SampX Code = 0
        Set the rest of the code:
            if ( Diff >= StepTab[ Index ] )
                SampX Code = SampX Code | 4
                Diff = Diff - StepTab[ Index ]
            if ( Diff >= ( StepTab[ Index ] >> 1 )
                SampX Code = SampX Code | 2
                Diff = Diff - ( StepTab[ Index ] >> 1 )
            if ( Diff >= ( StepTab[ Index ] >> 2 )
                SampX Code = SampX Code | 1
    Save the sample code, SampX Code            in the block
    Predict the current sample based on the sample code:
        Calculate the difference:
            Diff = 0
            if ( SampX Code & 4 )
                Diff = Diff + StepTab[ Index ]
            if ( SampX Code & 2 )
                Diff = Diff + ( StepTab[ Index ] >> 1 )
            if ( SampX Code & 1 )
                Diff = Diff + ( StepTab[ Index ] >> 2 )
            Diff = Diff + ( StepTab[ Index ] >> 3 )
        Check the sign bit:
            if ( SampX Code & 8 )
                Diff = -Diff
        SampX = SampX-1 + Diff
    Check for overflow and underflow errors:
        if PredSamp too large, make it the maximum allowable size (32767)
        if PredSamp too small, make it the minimum allowable size (-32768)
    Adjust the step table index:
        Index = Index + IndexTab[ SampX Code ]
    Check for step table index overflow and underflow:
        if Index too large, make it the maximum allowable size (88)
        if Index too small, make it the minimum allowable size (0)
```

This section describes the algorithm used for encoding the 3 bit DVI ADPCM. This procedure must be followed for each block for each channel.

```
For the first block only, clear the initial step table index:
    Index = 0

Get the first sample, Samp0
```

```
Create the block header:
    Write the first sample, Samp0, to the header
    Write the initial step table index, Index, to the header
Set the previously predicted sample value:
    PredSamp = Samp0
While there are still samples to encode, and we're not at the end of the block
    Get the next sample to encode, SampX
    Calculate the new sample code:
        Diff = SampX - PredSamp
        Set the sign bit:
            if ( Diff < 0 )
                SampX Code = 4
                Diff = -Diff
            else
                SampX Code = 0
        Set the rest of the code:
            if ( Diff >= StepTab[ Index ] )
                SampX Code = SampX Code | 2
                Diff = Diff - StepTab[ Index ]
            if ( Diff >= ( StepTab[ Index ] >> 1 )
                SampX Code = SampX Code | 1
    Save the sample code, SampX Code        in the block
    Predict the current sample based on the sample code:
        Calculate the difference:
            Diff = 0
            if ( SampX Code & 2 )
                Diff = Diff + StepTab[ Index ]
            if ( SampX Code & 1 )
                Diff = Diff + ( StepTab[ Index ] >> 1 )
            Diff = Diff + ( StepTab[ Index ] >> 2 )
        Check the sign bit:
            if ( SampX Code & 4 )
                Diff = -Diff
        SampX = SampX-1 + Diff
    Check for overflow and underflow errors:
        if PredSamp too large, make it the maximum allowable size (32767)
        if PredSamp too small, make it the minimum allowable size (-32768)
    Adjust the step table index:
        Index = Index + IndexTab[ SampX Code ]
    Check for step table index overflow and underflow:
        if Index too large, make it the maximum allowable size (88)
        if Index too small, make it the minimum allowable size (0)
```

DSP Solutions formerly Digispeech Wave Types

Added: 05/22/92
Author: Digispeech

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_DIGISTD (0x0015)
#define WAVE_FORMAT_DIGIFIX (0x0016)

wFormatTag	This must be set to either WAVE_FORMAT_DIGISTD or WAVE_FORMAT_DIGIFIX.
nChannels	Number of channels in the wave. (1 for mono)
nSamplesPerSec	Frequency the sample rate of the wave file. (8000). This value is also used by the fact chunk to determine the length in time units of the date.
nAvgBytesPerSec	Average data rate. (1100 for DIGISTD or 1625 for DigiFix)
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.
nBlockAlign	Block Alignment of 2 for DIGISTD and 26 for DigiFix.
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be zero.

The definition of the data contained in the Digistd and DigiFix formats are considered proprietary information of Digispeech. They can be contacted at:

DSP Solutions, Inc.
2464 Embarcadero Way
Palo Alto, CA 94303

The DIGISTD is a format used in a compression technique developed by Digispeech, Inc. DIGISTD format provides good speech quality with average rate of about 1100 bytes/second. The blocks (or buffers) in this format cannot be cyclically repeated.

The DigiFix is a format used in a compression technique developed by Digispeech, Inc. DigiFix format provides good speech quality (similar to DIGISTD) with average rate of exactly 1625 bytes/second. This format uses blocks of 26 bytes long.

Yamaha ADPCM

Added 09/25/92
Author: Yamaha

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_YAMAHA_ADPCM (0x0020)

wFormatTag	This must be set to WAVE_FORMAT_YAMAHA_ADPCM.	
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.	
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 5125, 7350, 9600, 11025, 22050, or 44100 Hz. Other sample rates are not allowed.	
nAvgBytesPerSec	Average data rate..	
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.	
nBlockAlign	This is dependent upon the number of bits per sample.	
	wBitsPerSample	nBlockAlign
	4	1
	4	1
wBitsPerSample	This is the number of bits per sample of YADPCM. Currently only 4 bits per sample is defined. Other values are reserved.	
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be zero.	

This format is created and read by Yamaha chip included in the Gold Sound Standard (GSS) that is implemented in a number of manufacturers boards. The algorithm and conversion routines are published in the source code provided in YADPCM.C with this technote.

Sonarc™ Compression

Added 10/21/92
Author: Sound Compression

Sound Compression has developed a new compression algorithm which, unlike ADPCM, is capable of lossless compression of digitized audio files to a degree far greater (50-60%) than that achievable with the other compressors, PKZIP and LHarc. "Lossy" compression is possible with even higher ratios. Information about the algorithm is available form the address below.

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

```
typedef struct sonarcwaveformat_tag {
      WAVEFORMATEX      wfx;
      WORD              wCompType;
} SONARCWAVEFORMAT
```

#define WAVE_FORMAT_SONARC (0x0021)

wFormatTag	This must be set to WAVE_FORMAT_SONARC.
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 11025, 22050, or 44100 Hz. Other sample rates are not allowed.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.
nBlockAlign	The valid values have not been defined.
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of SONARC.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 2.
wCompType	This value is not yet defined..

"Sonarc" is a trademark of Speech Compression.

To get information on this format please contact:

Speech Compression
1682 Langley Ave.
Irvine, CA 92714
Telephone: 714-660-7727 Fax: 714-660-7155

Creative Labs ADPCM

Added 10/01/92
Author: Creative Labs

Createive has defined a new ADPCM compression scheme, and this new scheme will be implemented on their H/W and will be able to support compression and decompression real-time. They do not provide a description of this algorithm. Information about the algorithm is available form the address below.

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

```
typedef struct creative_adpcmwaveformat_tag {
      WAVEFORMATEX      wfx;
      WORD              wRevision;
} CREATIVEADPCMWAVEFORMAT
```

#define WAVE_FORMAT_CREATIVE_ADPCM (0x0200)

wFormatTag	This must be set to WAVE_FORMAT_CREATIVE_ADPCM.		
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.		
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 8000, 11025, 22050, or 44100 Hz. Other sample rates are not allowed.		
nAvgBytesPerSec	Average data rate..		
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.		
nBlockAlign	This is dependent upon the number of bits per sample.		
	wBitsPerSample	nChannels	nBlockAlign
	4	1	1
	4	2	1
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.		
wBitsPerSample	This is the number of bits per sample of CADPCM.		
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 2.		
wRevision	Revision of algorithm. This should be one for the current definition.		

To get information on this format please contact:

Creative Developer Support
1901, McCarthy Blvd, Milpitas, CA 95035.
Tel : 408-428 6644 Fax : 408-428 6655

DSP Group Wave Type

Added: 01/04/93
Author: Paul Beard, DSP Group

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_DSPGROUP_TRUESPEECH (0x0022)

wFormatTag	This must be set to WAVE_FORMAT_DSPGROUP_TRUESPEECH.
nChannels	Number of channels in the wave, 1 for mono.
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 8000
nAvgBytesPerSec	Average data rate.. (1067)
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.
nBlockAlign	This is the block alignment of the data in bytes. (32).
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of TRUESPEECH. Not used; set to zero.
cbExtraSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 32.
wRevision	Revision no (1,...)
nSamplesPerBlock	Number of samples per block. 240

<nSamplesPerBlock> = <nSamplesPerSec> / <nAvgBytesPerSec> * <nBlockAlign>)

The definition of the data contained in the TRUESPEECH format is considered proprietary information of DSP Group Inc. They can be contacted at:

DSP Group Inc.,
4050 Moorpark Ave.,
San Jose CA. 95117
(408) 985 0722

TRUESPEECH is a format used in a compression technique developed by DSP Group Inc. TRUESPEECH format provides high quality telephony bandwidth voice vocoding with a rate of 1067 bytes per second. This format uses blocks of 32 bytes long.

Echo Speech Wave Type

Added: 01/21/93
Author: Echo Speech Corporation

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_ECHOSC1 (0x0023)

wFormatTag	This must be set to WAVE_FORMAT_ECHOSC1.
nChannels	Number of channels in the wave, always 1 for mono.
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 11025
nAvgBytesPerSec	Average data rate.. (450)
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.
nBlockAlign	This is the block alignment of the data in bytes. (6).
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample. Not used; set to zero.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 0.

The definition of the data contained in the ECHO SC-1 format is considered proprietary information of Echo Speech Corporation. They can be contacted at:

Echo Speech Corporation
6460 Via Real
Carpinteria, CA. 93013
805 684-4593

ECHO SC-1 is a format used in a compression technique developed by Echo Speech Corporation. ECHO SC-1 format provides excellent speech quality with an average data rate of exactly 450 bytes/second. This format uses blocks 6 bytes long.

ECHO is a registered trademark of Echo Speech Corporation.

AUDIOFILE Wave Type AF36

Added: April 29, 1993
Author: AudioFile

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_AUDIOFILE_AF36 (0x0024)

wFormatTag	This must be set to WAVE_FORMAT_AUDIOFILE_AF36
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

Audio File AF36 format provides very high compression for speech -based waveform audio. (Relative to 11 kHz, 16-bit PCM, a compression ratio of 36-to-1 is achieved with AF36.

For more information on AF36 and other AudioFile host-based and DSP based compression software contact: :

AudioFile, Inc.
Four Militia Drive
Lexington, MA, 02173
(617) 861-2996

Comment

Trademark info.

Audio Processing Technology Wave Type

Added: 06/22/93
Author: Calypso Software Limited

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_APTX (0x0025)

wFormatTag	This must be set to WAVE_FORMAT_APTX.
nChannels	Number of channels in the wave, always 1 for mono, 2 for stereo.
nSamplesPerSec	Frequency of the sample rate of the wave file. (8000, 11025, 22050, 44100, 48000)
nAvgBytesPerSec	Average data rate..= nChannels * nSamplesPerSec/2. (16bit audio)
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.
nBlockAlign	Should be set to 2 (bytes) for mono data or 4 (bytes) for stereo. For mono data 4 sixteen bit samples will be compressed into 1 sixteen bit word For stereo data 4 sizteen bit left channel samples will be compressed into the first 16bit word and 4 sixteen bit right channel samples will be cmpressed into the next 16 bit word.
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample. Not used; set to four.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 0.(zero)

The definition of the data contained in the APTX format is considered proprietary information of Audio Processing Technology Limited. They can be contacted at:

Audio Processing Technology Limited
Edgewater Road
Belfast, Northern Ireland, BT3 9QJ
Tel 44 232 371110
Fax 44 232 371137

This format is proprietary audio format using 4:1 compression i.c. 16 bits of audio are compressed to 4 bits. It is only encoded/decoded by dedicated hardware from MM_APT

AUDIOFILE Wave Type AF10

Added: June 22, 1993
Author: AudioFile

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_AUDIOFILE_AF10 (0x0026)

wFormatTag	This must be set to WAVE_FORMAT_AUDIOFILE_AF10
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

For more information on AF36 and other AudioFile host-based and DSP based compression software contact: :

AudioFile, Inc.
Four Militia Drive
Lexington, MA, 02173
(617) 861-2996

Dolby Labs AC-2 Wave Type

Added: 06/24/93
Author: Dolby Laboratories, Inc.

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_DOLBY_AC2 (0x0030)

wFormatTag	This must be set to WAVE_FORMAT_DOLBY_AC2
nChannels	Number of channels, 1 for mono, 2 for stereo
nSamplesPerSec	Three sample rates allowed: 48000, 44100, 32000 samples per second
nAvgBytesPerSec	Average data rate. ((nSamplesperSec*nBlockAlign)/512
nBlockAlign	The block alignment (in bytes) of the dat in <data-ck>. Given in table
nSamplesPerSec	nBlockAlign
48000	nChannels*168
44100	nChannels*184
32000	nChannels*190
wBitsPerSample	Approximately 3 bits per sample
cbExtraSize	2 extra bytes of information in format header
nAuxBitsCode	Auxiliary bits code indicating number of Aux. bits per block. The amount of audio data bits is reduced by this number in the decoder, such that the overall block size remains constant.
nAuxBitsCode	Number of Aux bits in block
0	0
1	8
2	16
3	32

specific structure of the **<wave-data>** chunk is proprietary, and may be obtained from Dolby Laboratories. Also contact Dolby for methods of including **<assoc-data-list>** chunks.

Dolby Laboratories
100 Potrero Avenue
San Francisco, CA 94103-4813
Tel 415-558-0200

```
/* Dolby's AC-2 wave format structure definition */
typedef struct dolbyac2waveformat_tag {
        WAVEFORMATEX    wfx;
        WORD            nAuxBitsCode;
} DOLBYAC2WAVEFORMAT;
```

Sierra ADPCM

Added 07/26/93
Author: Sierra Semiconductor Corp.

Sierra Semiconductor has developed a compression scheme similar to the standard CCITT ADPCM. This scheme has been implemented in Aria™-based sound boards and is capable of supporting compression and decompression in real-time. A description of the algorithm is not available at this time.

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

```
typedef struct sierra_adpcmwaveformat_tag {
      EXTWAVEFORMAT     ewf;
      WORD              wRevision;
} SIERRAADPCMWAVEFORMAT;
```

#define WAVE_FORMAT_SIERRA_ADPCM (0x0013)

wFormatTag	This must be set to WAVE_FORMAT_SIERRA_ADPCM.		
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.		
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 22050 Hz. Other sample rates are not currently allowed.		
nAvgBytesPerSec	Average data rate.		
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.		
nBlockAlign	This is dependent upon the number of bits per sample.		
	wBitsPerSample	nChannels	nBlockAlign
	4	1	1
	4	2	1
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.		
wBitsPerSample	This is the number of bits per sample of Sierra ADPCM. Currently, only 4 bits per sample is defined. Other values are reserved.		
cbExtraSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 2.		
wRevision	Revision of algorithm. This should be 0x0100 for the current definition.		

VideoLogic Wave Types

Added: 07/13/93
Author: VideoLogic

Fact Chunck

Wave Format Header

#define WAVE_FORMAT_MEDIASPACE_ADPCM (0x0012)

```
//
//  VideoLogic's MediaSpace ADPCM structure definitions
//
//     for WAVE_FORMAT_MEDIASPACE_ADPCM   (0x0012)
//
//

typedef struct mediaspace_adpcmwaveformat_tag {
      WAVEFORMATEX      wfx;
      WORD        wRevision;
} MEDIASPACEADPCMWAVEFORMAT;
typedef MEDIASPACEADPCMWAVEFORMAT       *PMEDIASPACEADPCMWAVEFORMAT;
typedef MEDIASPACEADPCMWAVEFORMAT NEAR *NPMEDIASPACEADPCMWAVEFORMAT;
typedef MEDIASPACEADPCMWAVEFORMAT FAR  *LPMEDIASPACEADPCMWAVEFORMAT;
```

CCITT G.723 ADPCM

Added: 08/25/93
Author: Antex Electronics Corp.

The algorithm for G.721 header format is essentially the same as G723.

Fact Chunk

WAVE Format Header

#define WAVE_FORMAT_G723_ADPCM (0x0014)

wFormatTag	This must be set to WAVE_FORMAT_G.723_ADPCM		
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo		
nSamplesPerSec	Frequency the sample rate of the wave file. (8000, 11025, 22050, 44100)		
nAvgBytesPerSec	Average data rate		
	Playback software can estimate the buffer size using the <nAvgBytesPerSec> value.		
nBlockAlign	This is dependent upon the number of bits per sample.		
	wBitsPerSample	nChannels	nBlockAlign
	3	1	48 + nAuxBlockSize
	3	2	96 + nAuxBlockSize
	5	1	80 + nAuxBlockSize
	5	2	160 + nAuxBlockSize
	Playback software needs to process a multiple of <nBlockAlign> bytes of data at a time, so that the value of <nBlockAlign> can be used for buffer alignment.		
wBitsPerSample	This is the number of bits per sample of data. (G.723 can be 3 or 5)		
cbExtraSize	The size in bytes of the extra information in the extended WAVE 'fmt' header. This should be 2.		
nAuxBlockSize	This is the size in bytes of auxiliary data that is stored at the beginning of each data block. In most instances this should be set to 0.		

See the G.723 specification for algorithm details.

Data Format

Mono, 3 bits per sample
Grouped into 3 byte sub-blocks containing 8 mono samples. The bit ordering for samples labeled A through H is:

<A2|A1|A0|B2|B1|B0|C2|C1> <C0|D2|D1|D0|E2|E1|E0|F2> <F1|F0|G2|G1|G0|H2|H1|H0>
Byte 1 Byte 2 Byte 3

;where A2 is the MSB and A0 is the LSB of the first sample.

Stereo, 3 bits per sample
Grouped into 6 byte sub-blocks containing 8 stereo samples. The bit ordering for samples labeled A through H is:

<AL2|AL1|AL0|AR2|AR1|AR0|BL2|BL1> <BL0|BR2|BR1|BR0|CL2|CL1|CL0|CR2>
Byte 1 Byte 2

<CR1|CR0|DL2|DL1|DL0|DR2|DR1|DR0> <EL2|EL1|EL0|ER2|ER1|ER0|FL2|FL1>
Byte 3 Byte 4

<FL0|FR2|FR1|FR0|GL2|GL1|GL0|GR2> <GR1|GR0|HL2|HL1|HL0|HR2|HR1|HR0>
Byte 5 Byte 6

;where AL2 is the MSB and AL0 is the LSB of the first left sample, and AR2 is the MSB and AR0 is the LSB of the first right sample

Mono, 5 bits per sample
Grouped into 5 byte sub-blocks containing 8 mono samples. The bit ordering for samples labeled A through H is:

<A4|A3|A2|A1|A0|B4|B3|B2> <B1|B0|C4|C3|C2|C1|C0|D4>
<D3|D2|D1|D0|E4|E3|E2|E1>
Byte 1 Byte 2 Byte 3

<E0|F4|F3|F2|F1|F0|G4|G3> <G2|G1|G0|H4|H3|H2|H1|H0>
Byte 4 Byte 5

;where A4 is the MSB and A0 is the LSB of the first sample.

Stereo, 5 bits per sample
Grouped into 10 byte sub-blocks containing 8 stereo samples. The bit ordering for samples labeled A through H is:

<AL4|AL3|AL2|AL1|AL0|AR4|AR3|AR2> <AR1|AR0|BL4|BL3|BL2|BL1|BL0|BR4>
Byte 1 Byte 2

<BR3|BR2|BR1|BR0|CL4|CL3|CL2|CL1> <CL0|CR4|CR3|CR2|CR1|CR0|DL4|DL3>
Byte 3 Byte 4

<DL2|DL1|DL0|DR4|DR3|DR2|DR1|DR0> <EL4|EL3|EL2|EL1|EL0|ER4|ER3|ER2>
Byte 5 Byte 6

<ER1|ER0|FL4|FL3|FL2|FL1|FL0|FR4> <FR3|FR2|FR1|FR0|GL4|GL3|GL2|GL1>
Byte 7 Byte 8

<GL0|GR4|GR3|GR2|GR1|GR0|HL4|HL3> <HL2|HL1|HL0|HR4|HR3|HR2|HR1|HR0>
Byte 9 Byte 10

;where AL4 is the MSB and AL0 is the LSB of the first left sample, and AR4 is the MSB and AR0 is the LSB of the first right sample

Dialogic OKI ADPCM

Added: 04/07/94
Author: Dialogic

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_DIALOGIC_OKI_ADPCM (0x0203)

wFormatTag	This must be set to WAVE_FORMAT_DIALOGIC_OKI_ADPCM.
nChannels	Number of channels in the wave. 1
nSamplesPerSec	Frequency the of the sample rate of wave file. 6000, 8000,
nAvgBytesPerSec	Average data rate. 3000, 4000
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of for the data. 1
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data. 4
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. 0

This format can be created and read by either OKI ADPCM chip set of by a firmware program.

Control Resources Limited VQLPC

Added: 04/05/94
Author: Control Resources Limited

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_CONTROL_RES_VQLPC (0x0034)

wFormatTag	This must be set to WAVE_FORMAT_CONTROL_RES_VQLPC
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file. 8000
nAvgBytesPerSec	Average data rate.394
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data in Bytes. 18
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data. 4
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. 2
wCompType	This value is reserved and should be set to 1

VQLPC is trademarked of Control Resources Ltd.

Control Resources Limited CR10

Added: 04/05/94
Author: Control Resources Limited

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_CONTROL_RES_CR10 (0x0037)

wFormatTag	This must be set to WAVE_FORMAT_CONTROL_RES_CR10.
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

data not available at time of printing.

G.721 WAVE Format Header

Added: 08/25/93
Author: Antex Electronics Corp.

The algorithm for G.721 header format is essentially the same as G723.

Fact Chunk

WAVE Format Header

#define WAVE_FORMAT_G721_ADPCM (0x0040)

wFormatTag	This must be set to WAVE_FORMAT_G721_ADPCM.	
nChannels	Number of channels in the wave.(1 for mono, 2 for stereo)	
nSamplesPerSec	Frequency the of the sample rate of wave file.	
nAvgBytesPerSec	Average data rate.	
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.	
nBlockAlign	Block Alignment of the data.	
	nChannels	nBlockAlign
	1	64+nAuxBlockSize
	2	128+nAuxBlockSize
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.	
wBitsPerSample	This is the number of bits per sample of data. This should be 4.	
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. This should be 2.	
nAuxBlockSize	This is the size in bytes of auxiliary data that is stored at the beginning of each data block. In most instances this should be set to 0.	

See the G.721 specification for algorithm details.
This is a CCITT (International Telegraph and Telephone Consultative Committee) specification. Their address is:

Palais des Nations
CH-1211 Geneva 10, Switzerland
Phone: 22 7305111

Data Format

Mono, 4 bits per sample
Grouped into 1 byte sub-blocks containing 2 mono samples. The bit ordering for samples labeled A and B is:

<A3|A2|A1|A0|B3|B2|B1|B0>

;where A3 is the MSB and A0 is the LSB of the first sample and B3 is the MSB and B0 is the LSB of the second sample.

Stereo, 4 bits per sample
Grouped into 1 byte sub-blocks containing 1 stereo sample. The bit ordering for one stereo sample is:

<L3|L2|L1|L0|R3|R2|R1|R0>

;where L3 is the MSB and L0 is the LSB of the left sample, and R3 is the MSB and R0 is the LSB of the right sample

ADPCME WAVE Format Header

Added: 10/23/93
Author: Antex Electronics Corp.

Fact Chunk

WAVE Format Header

#define WAVE_FORMAT_ADPCME (0x0033)

wFormatTag	This must be set to WAVE_FORMAT_ADPCME.
nChannels	Number of channels in the wave.(1 for mono, 2 for stereo)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data, 1 for Mono, 2 for Stereo.
wBitsPerSample	This is the number of bits per sample of data. This should be 4.
cbExtraSize	0

Data Format

Mono nibbles are labelled M and left and right samples labelled L and R.
Mono ADPCME

```
<M0|M1> <M2|M3> <M4|M5> <M6|M7>
byte 0     byte 1  byte 2  byte 3
```

Stereo ADPCME

```
<L1|R1> <L0|R0> <L3|R3> <L2|R2>
byte 0     byte 1  byte 2  byte 3
```

Note: Stereo nibble ordering is deliberately different from the mono order.

GSM610 Wave Type

Added: 09/05/93
Author: Microsoft

Fact Chunk

WAVE Format Header

```
typedef struct gsm610waveformat_tag {
WAVEFORMATEX        wfx;
WORD                wSamplesPerBlock;
} GSM610WAVEFORMAT;
typedef GSM610WAVEFORMAT *PGSM610WAVEFORMAT;
typedef GSM610WAVEFORMAT NEAR *NPGSM610WAVEFORMAT;
typedef GSM610WAVEFORMAT FAR *LPGSM610WAVEFORMAT;

#define        WAVE_FORMAT_GSM610                    (0x0031)
```

wFormatTag	This must be set to WAVE_FORMAT_GSM610
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

DSP Solutions REAL Wave Type

Added 02/03/94
Author: DSP Solutions (formerly Digispeech)

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE FORMAT HEADER

The extended wave format structure is used to defined all non-PCM format wave data, and is described as follows in the include file *mmreg.h*:

```
/* general extended waveform format structure */
/* Use this for all NON PCM formats */
/* (information common to all formats) */
```

```
typedef struct waveformat_extended_tag {
   WORD    wFormatTag;        /* format type */
   WORD    nChannels;         /* number of channels (i.e. mono, stereo...) */
   DWORD   nSamplesPerSec;    /* sample rate */
   DWORD   nAvgBytesPerSec;   /* for buffer estimation */
   WORD    nBlockAlign;       /* block size of data */
   WORD    wBitsPerSample;    /* Number of bits per sample of mono data */
   WORD    cbSize;            /* The count in bytes of the extra size */} WAVEFORMATEX;
```

#define WAVE_FORMAT_DIGIREAL (0x0035)

wFormatTag	Must be set WAVE_FORMAT_DIGIREAL
nChannels	Number of channels in the wave, 1 for mono.
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 8000. Other sample rates are allowed, but not encouraged. This rate is also used by the sample size entry in the fact chunk to determine the length in time of the data.
nAvgBytesPerSec	Average data rate (1650).
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	The block alignment (in bytes) of the data in <**data-ck**> (13).
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample per sample of data. (2). Each channel is assumed to have the same sample resolution. If this field is not needed, then it should be set to zero.
cbSize	The size in bytes of the extra information in the extended WAVE 'fmt' header. This should be 0.

DSP Solutions ADPCM Wave Type

Added 02/03/94
Author: DSP Solutions (formerly Digispeech)

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVEFORMATEX

The extended wave format structure is used to defined all non-PCM format wave data, and is described as follows in the include file *mmreg.h*:

```
/* general extended waveform format structure */
/* Use this for all NON PCM formats */
/* (information common to all formats) */
typedef struct waveformat_extended_tag {
   WORD    wFormatTag;        /* format type */
   WORD    nChannels;         /* number of channels (i.e. mono, stereo...) */
   DWORD   nSamplesPerSec;    /* sample rate */
   DWORD   nAvgBytesPerSec;   /* for buffer estimation */
   WORD    nBlockAlign;       /* block size of data */
   WORD    wBitsPerSample;    /* Number of bits per sample of mono data */
   WORD    cbSize;            /* The count in bytes of the extra size */} WAVEFORMATEX;
```

#define WAVE_FORMAT_DIGIADPCM (0x0036)

wFormatTag	Must be set to WAVE_FORMAT_DIGIADPCM		
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo		
nSamplesPerSec	Frequency of the sample rate of the wave file. This should be 11025, 22050, or 44100. Other sample rates are allowed.		
nAvgBytesPerSec	Average data rate.		
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.		
nBlockAlign	The block alignment (in bytes) of the data in <**data-ck**>.		
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.		
	wBitsPerSample	nChannels	nBlockAlign
	3	1	3
	3	2	6
wBitsPerSample	This is the number of bits per sample per channel data. (3)		
cbSize	The size in bytes of the extra information in the WAVE format. Should be 0.		

MPEG-1 Audio (Audio-only)

Added 18/01/93
Author: Microsoft

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_MPEG (0x0050)

```
typedef struct mpeg1waveformat_tag {
      WAVEFORMATEX       wfx;
      WORD               fwHeadLayer;
      DWORD              dwHeadBitrate;
      WORD               fwHeadMode;
      WORD               fwHeadModeExt;
      WORD               wHeadEmphasis;
      WORD               fwHeadFlags;
      DWORD              dwPTSLow;
      DWORD              dwPTSHigh;
} MPEG1WAVEFORMAT;
```

wFormatTag	This must be set to WAVE_FORMAT_MPEG.
nChannels	Number of channels in the wave, 1 for mono, 2 for stereo.

nSamplesPerSec	Sampling frequency (in Hz) of the wave file: 32000, 44100, or 48000. Note, however, that if the sampling frequency of the data is variable, then this field should be set to zero. It is *strongly* recommended that a fixed sampling frequency be used for desktop applications.
nAvgBytesPerSec	Average data rate; this might not be a legal MPEG bit rate if variable bit rate coding under layer 3 is used.
nBlockAlign	The block alignment (in bytes) of the data in <data-ck>. For audio streams which have a fixed audio frame length, the block alignment is equal to the length of the frame. For streams in which the frame length varies, nBlockAlign should be set to 1.
	With a sampling frequency of 32 or 48 kHz, the size of an MPEG audio frame is a function of the bit rate. If an audio stream uses a constant bit rate, the size of the audio frames does not vary. Therefore, the following formulas apply: Layer 1: nBlockAlign = 4*(int)(12*BitRate/SamplingFreq) Layers 2 and 3: nBlockAlign = (int)(144*BitRate/SamplingFreq) Example 1: For layer 1, with a sampling frequency of 32000 Hz and a bit rate of 256 kbits/s, nBlockAlign = 384 bytes.
	If an audio stream contains frames with different bit rates, then the length of the frames varies within the stream. Variable frame lengths also occur when using a sampling frequency of 44.1 kHz: in order to maintain the data rate at the nominal value, the size of an MPEG audio frame is periodically increased by one "slot" (4 bytes in layer 1, 1 byte in layers 2 and 3) as compared to the formulas given above. In these two cases, the concept of block alignment is invalid. The value of nBlockAlign must therefore be set to 1, so that MPEG-aware applications can tell whether the data is block-aligned or not. Note that it is possible to construct an audio stream which has constant-length audio frames at 44.1 kHz by setting the *padding_bit* in each audio frame header to the same value (either 0 or 1). Note, however, that bit rate of the resulting stream will not correspond exactly to the nominal value in the frame header, and therefore some decoders may not be capable of decoding the stream correctly. In the interested of standardization and compatibility, this approach is discouraged.
wBitsPerSample	Not used; set to zero.
cbSize	The size in bytes of the extended information after the WAVEFORMATEX structure. For the standard WAVE_FORMAT_MPEG format, this is 22. If extra fields are added, this value will increase.
fwHeadLayer	The MPEG audio layer, as defined by the following flags: ACM_MPEG_LAYER1 - layer 1. ACM_MPEG_LAYER2 - layer 2. ACM_MPEG_LAYER3 - layer 3. Some legal MPEG streams may contain frames of different layers. In this case, the above flags should be ORed together so that a driver may tell which layers are present in the stream.
dwHeadBitrate	The bit rate of the data, in bits per second. This value must be a standard bit rate according to the MPEG specification; not all bit rates are valid for all modes and layers. See Tables 1 and 2, below. Note that this field records the actual bit rate, not MPEG frame header code. If the bitrate is variable, or if it is a non-standard bit rate, then this field should be set to zero. It is recommended that variable bit rate coding be avoided where possible.

fwHeadMode	Stream mode, as defined by the following flags: ACM_MPEG_STEREO - stereo. ACM_MPEG_JOINTSTEREO - joint-stereo. ACM_MPEG_DUALCHANNEL - dual-channel (for example, a bilingual stream). ACM_MPEG_SINGLECHANNEL - single channel. Some legal MPEG streams may contain frames of different modes. In this case, the above flags should be ORed together so that a driver may tell which modes are present in the stream. This situation is particularly likely with joint-stereo encoding, as encoders may find it useful to switch dynamically between stereo and joint-stereo according to the characteristics of the signal. In this case, both the ACM_MPEG_STEREO and the ACM_MPEG_JOINTSTEREO flags should be set.
fwHeadModeExt	Contains extra parameters for joint-stereo coding; not used for other modes. See Table 3, below. Some legal MPEG streams may contain frames of different mode extensions. In this case, the values in Table 3 may be ORed together. Note that fwHeadModeExt is only used for joint-stereo coding; for other modes (single channel, dual channel, or stereo), it should be set to zero. In general, encoders will dynamically switch between the various possible *mode_extension* values according to the characteristics of the signal. Therefore, for normal joint-stereo encoding, this field should be set to 0x000f. However, if it is desirable to limit the encoder to a particular type of joint-stereo coding, this field may be used to specify the allowable types.
wHeadEmphasis	Describes the de-emphasis required by the decoder; this implies the emphasis performed on the stream prior to encoding. See Table 4, below.
fwHeadFlags	Sets the corresponding flags in the audio frame header: ACM_MPEG_PRIVATEBIT - set the private bit. ACM_MPEG_COPYRIGHT - set the copyright bit. ACM_MPEG_ORIGINALHOME - sets the original/home bit. ACM_MPEG_PROTECTIONBIT - sets the protection bit, and inserts a 16-bit error protection code into each frame. ACM_MPEG_ID_MPEG1 - sets the ID bit to 1, defining the stream as an MPEG-1 audio stream. *This flag must always be set explicitly to maintain compatibility with future MPEG audio extensions (i.e. MPEG-2).* An encoder will use the value of these flags to set the corresponding bits in the header of each MPEG audio frame. When describing an encoded data stream, these flags represent a logical OR of the flags set in each frame header. That is, if the copyright bit is set in one or more frame headers in the stream, then the ACM_MPEG_COPYRIGHT flag will be set. Therefore, the value of these flags is not necessarily valid for every audio frame.
dwPTSLow	This field (together with the following field) consists of the presentation time stamp (PTS) of the first frame of the audio stream, as taken from the MPEG system layer. dwPTSLow contains the 32 LSBs of the 33-bit PTS. The PTS may be used to aid in the re-integration of an audio stream with an associated video stream. If the audio stream is not associated with a system layer, then this field should be set to zero.
dwPTSHigh	This field (together with the previous field) consists of the presentation time stamp (PTS) of the first frame of the audio stream, as taken from the MPEG system layer. The LSB of dwPTSHigh contains the MSB of the 33-bit PTS. The PTS may be used to aid in the re-integration of an audio stream with an associated video stream. If the audio stream is not associated with a system layer, then this field should be set to zero.
	Note: The previous two fields can be treated as a single 64-bit integer; optionally, the dwPTSHigh field can be tested as a flag to determine whether the MSB is set or cleared.

Table 1: Allowable Bit Rates (bits/s)

MPEG frame header code	Layer 1	Layer 2	Layer 3
'0000'	free format	free format	free format
'0001'	32000	32000	32000
'0010'	64000	48000	40000
'0011'	96000	56000	48000
'0100'	128000	64000	56000
'0101'	160000	80000	64000
'0110'	192000	96000	80000
'0111'	224000	112000	96000
'1000'	256000	128000	112000
'1001'	288000	160000	128000
'1010'	320000	192000	160000
'1011'	352000	224000	192000
'1100'	384000	256000	224000
'1101'	416000	320000	256000
'1110'	448000	384000	320000
'1111'	forbidden	forbidden	forbidden

Table 2: Allowable mode-bitrate combinations for Layer 2.

Bit rate (bits/sec)	Allowable modes
32000	single channel
48000	single channel
56000	single channel
64000	all modes
80000	single channel
96000	all modes
112000	all modes
128000	all modes
160000	all modes
192000	all modes
224000	stereo, intensity stereo, dual channel
256000	stereo, intensity stereo, dual channel
320000	stereo, intensity stereo, dual channel
384000	stereo, intensity stereo, dual channel

Table 3: Mode Extension

fwHeadModeExt	MPEG frame header code	Layers 1 and 2	Layers 3
0x0001	'00'	subbands 4-31 in intensity stereo	no intensity or ms-stereo coding
0x0002	'01'	subbands 8-31 in intensity stereo	intensity stereo

0x0004	'10'	subbands 12-31 in intensity stereo	ms-stereo
0x0008	'11'	subbands 16-31 in intensity stereo	both intensity and ms-stereo coding

Table 4: Emphasis Field

wHeadEmphasis	MPEG frame header code	De-emphasis required
1	'00'	no emphasis
2	'01'	50/15 ms emphasis
3	'10'	reserved
4	'11'	CCITT J.17

Flags

The following flags are defined for the **fwHeadLayer** field. For encoding, one of these flags should be set so that the encoder knows what layer to use. For decoding, the driver can check these flags to determine whether it is capable of decoding the stream. Note that a legal MPEG stream may use different layers in different frames within a single stream. Therefore, more than one of these flags may be set.

```
#define ACM_MPEG_LAYER1                  (0x0001)
#define ACM_MPEG_LAYER2                  (0x0002)
#define ACM_MPEG_LAYER3                  (0x0004)
```

The following flags are defined for the **fwHeadMode** field. For encoding, one of these flags should be set so that the encoder knows what layer to use; for joint-stereo encoding, typically the ACM_MPEG_STEREO and ACM_MPEG_JOINTSTEREO flags will both be set so that the encoder can use joint-stereo coding only when it is more efficient than stereo. For decoding, the driver can check these flags to determine whether it is capable of decoding the stream. Note that a legal MPEG stream may use different layers in different frames within a single stream. Therefore, more than one of these flags may be set.

```
#define ACM_MPEG_STEREO                  (0x0001)
#define ACM_MPEG_JOINTSTEREO             (0x0002)
#define ACM_MPEG_DUALCHANNEL             (0x0004)
#define ACM_MPEG_SINGLECHANNEL           (0x0008)
```

Table 3 defines flags for the **fwHeadModeExt** field. This field is only used for joint-stereo coding; for other encoding modes, this field should be set to zero. For joint-stereo encoding, these flags indicate the types of joint-stereo encoding which an encoder is permitted to use. Normally, an encoder will dynamically select the mode extension which is most appropriate for the input signal; therefore, an application would typically set this field to 0x000f so that the encoder may select between all possibilities; however, it is possible to limit the encoder by clearing some of the flags. For an encoded stream, this field indicates the values of the MPEG *mode_extension* field which are present in the stream.

The following flags are defined for the **fwHeadFlags** field. These flags should be set before encoding so that the appropriate bits are set in the MPEG frame header. When describing an encoded MPEG audio stream, these flags represent a logical OR of the corresponding bits in the header of each audio frame. That is, if the bit is set in any of the frames, it is set in the **fwHeadFlags** field. If an application wraps a RIFF WAVE header around a pre-encoded MPEG audio bit stream, it is responsible for parsing the bit stream and setting the flags in this field.

```
#define ACM_MPEG_PRIVATEBIT          (0x0001)
#define ACM_MPEG_COPYRIGHT           (0x0002)
#define ACM_MPEG_ORIGINALHOME        (0x0004)
#define ACM_MPEG_PROTECTIONBIT       (0x0008)
#define ACM_MPEG_ID_MPEG1            (0x0010)
```

Data

The data chunk consists of an MPEG-1 audio sequence as defined by the ISO 11172 specification, part 3 (audio). This sequence consists of a bit stream, which is stored in the data chunk as an array of bytes. Within a byte, the MSB is the first bit of the stream, and the LSB is the last bit. The data is *not* byte-reversed. For example, the following data consists of the first 16 bits (from left to right) of a typical audio frame header:

```
Syncword      ID  Layer  ProtectionBit ...
111111111111  1   10     1             ...
```

This data would be stored in bytes in the following order:

```
Byte0  Byte1 ...
FF     FD    ...
```

MPEG Audio Frames

An MPEG audio sequence consists of a series of audio frames, each of which begins with a frame header. Most of the fields within this frame header correspond to fields in the MPEG1WAVEFORMAT structure defined above. For encoding, these fields can be set in the MPEG1WAVEFORMAT structure, and the driver can use this information to set the appropriate bits in the frame header when it encodes. For decoding, a driver can check these fields to determine whether it is capable of decoding the stream.

Encoding

A driver which encodes an MPEG audio stream should read the header fields in the MPEG1WAVEFORMAT structure and set the corresponding bits in the MPEG frame header. If there is any other information which a driver requires, it must get this information either from a configuration dialog box, or through a driver callback function. For more information, see the Ancillary Data section, below.

If a pre-encoded MPEG audio stream is wrapped with a RIFF header, it is the responsibility of the application to parse the bit stream and set the fields in the MPEG1WAVEFORMAT structure. If the sampling frequency or the bitrate index is not constant throughout the data stream, the driver should set the corresponding MPEG1WAVEFORMAT fields (**nSamplesPerSec** and **dwHeadBitrate**) to zero, as described above. If the stream contains frames of more than one layer, it should set the flags in **fwHeadLayer** for all layers which are present in the stream. Since fields such as **fwHeadFlags** can vary from frame to frame, caution must be used in setting and testing these flags; in general, an application should not rely on them to be valid for every frame. When setting these flags, adhere to the following guidelines:

- ACM_MPEG_COPYRIGHT should be set if any of the frames in the stream have the copyright bit set.
- ACM_MPEG_PROTECTIONBIT should be set if any of the frames in the stream have the protection bit set.

- ACM_MPEG_ORIGINALHOME should be set if any of the frames in the stream have the original/home bit set. This bit may be cleared if a copy of the stream is made.
- ACM_MPEG_PRIVATEBIT should be set if any of the frames in the stream have the private bit set.
- ACM_MPEG_ID_MPEG1 should be set if any of the frames in the stream have the ID bit set. For MPEG-1 streams, the ID bit should always be set; however, future extensions of MPEG (such as the MPEG-2 multi-channel format) may have the ID bit cleared.

If the MPEG audio stream was taken from a system-layer MPEG stream, or if the stream is intended to be integrated into the system layer, then the presentation time stamp (PTS) fields may be used. The PTS is a field in the MPEG system layer which is used for synchronization of the various fields. The MPEG PTS field is 33 bits, and therefore the RIFF WAVE format header stores the value in two fields: **dwPTSLow** contains the 32 LSBs of the PTS, and **dwPTSHigh** contains the MSB. These two fields may be taken together as a 64-bit integer; optionally, the **dwPTSHigh** field may be tested as a flag to determine whether the MSB is set or cleared. When extracting an audio stream from a system layer, a driver should set the PTS fields to the PTS of the first frame of the audio data. This may later be used to re-integrate the stream into the system layer. *The PTS fields should not be used for any other purpose.* If the audio stream is not associated with the MPEG system layer, then the PTS fields should be set to zero.

Decoding

A driver may test the fields in the MPEG1WAVEFORMAT structure to determine whether it is capable of decoding the stream. However, the driver must be aware that some fields, such as the **fwHeadFlags** field, may not be consistent for every frame in the bit stream. A driver should never use the fields of the MPEG1WAVEFORMAT structure to perform the actual decoding. The decoding parameters should be taken entirely from the MPEG data stream.

A driver may check the **nSamplesPerSec** field to determine whether it supports the sampling frequency specified. If the MPEG stream contains data with a variable sampling rate, then the **nSamplesPerSec** field will be set to zero. If the driver cannot handle this type of data stream, then it should not attempt to decode the data, but should fail immediately.

Ancillary Data

The audio data in an MPEG audio frame may not fill the entire frame. Any remaining data is called *ancillary data*. This data may have any format desired, and may be used to pass additional information of any kind. If a driver wishes to support the ancillary data, it must have a facility for passing the data to and from the calling application. The driver may use a callback function for this purpose. Basically, the driver may call a specified callback function whenever it has ancillary data to pass to the application (i.e. on decode) or whenever it requires more ancillary data (on encode).

Drivers should be aware that not all applications will want to process the ancillary data. Therefore, a driver should only provide this service when explicitly requested by the application. The driver may define a custom message which enables and disables the callback facility. Separate messages could be defined for the encoding and decoding operations for more flexibility.

If the callback facility is enabled, then the application is responsible for creating a callback function which is capable of processing the ancillary data. Typically, the application already has a callback defined in order to feed data blocks to the wave device as they are needed; this callback processes the WOM_CLOSE, WOM_DONE, and WOM_OPEN messages, and/or the WIM_CLOSE, WIM_DATA, and WIM_OPEN messages. The address of the callback function (or a window handle) is passed to the driver by the waveOutOpen or the waveInOpen calls in the **dwCallback** parameter. Two additional messages must defined by the driver and supported by the callback: one to pass ancillary data back to the application (i.e. WOM_ANCDATA_OUT), and one to request ancillary data from the application (i.e. WIM_ANCDATA_IN).

As message parameters, the WOM_ANCDATA_OUT could pass a pointer to a data buffer, and a size parameter indicating the number of bits (or bytes) of data in the buffer. The buffer would be allocated by the driver, and freed after the message has been processed by the callback. The driver could pass back the ancillary data frame by frame as it is received, or it could process an entire block of data and pass back the ancillary data in a single large chunk. The method is up to the driver, or could be configurable either through a configuration dialog or as a parameter passed when the ancillary data functions are enabled by the application.

To request ancillary data, the WIM_ANCDATA_IN message could pass a pointer to an empty data buffer, which the callback function would fill with ancillary data. If the amount of ancillary data varies from frame to frame, the first few bytes of the buffer could be defined to be the number of bits (or bytes) of data. This buffer would be allocated and freed by the driver; in order to ensure that there is enough space to hold all the data, the buffer size could be configurable using either a configuration dialog or by passing the value to the driver as a parameter when the ancillary data functions are enabled by the application.

Note that this method may not be appropriate for all drivers or all applications; it is included only as an illustration of how ancillary data may be supported. For more information, consult the Windows 3.1 Software Development Kit, "Multimedia Programmers Reference," and the Windows 3.1 Device Driver Kit, "Multimedia Device Adaptation Guide."

Standards

It is recommended that applications use the 44.1 kHz sampling rate whenever possible, to maintain compatibility with current computer standards. It is also recommended that encoders avoid the use of variable bitrate coding, and it is *strongly* recommended that all bit streams use a constant sampling frequency. Streams which have a variable sampling frequency cannot be decoded to PCM for manipulation by other audio services.

References

ISO/IEC JTC1/SC29/WG11 MPEG, April 1992. ISO/IEC Draft International Standard: "Coding of moving pictures and associated audio for digital storage media up to about 1.5 Mbit/s."

Creative Labs, Inc. FastSpeech 8 & 10

Added: 03/2/94
Author: Creative Labs

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_CREATIVE_FASTSPEECH8 (0x0202)

#define WAVE_FORMAT_CREATIVE_FASTSPEECH10 (0x0203)

wFormatTag	This must be set to WAVE_FORMAT_CREATIVE_FASTSPEECH8 or 10
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file. 8000 or 11025
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of for the data. 32 for FASTSPEECH8 and 26 for FASTSPEECH10
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbExtraSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. 2.
wRevision	Revision of the Algorithm. This should be 1 for the current definition.

Fujitsu FM Towns SND Wave Type

Added: 02/15/94
Author: Fujitsu

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_FM_TOWNS_SND (0x0300)

wFormatTag	This must be set to WAVE_FORMAT_FM_TOWNS_SND
nChannels	Number of channels in the wave. 1
nSamplesPerSec	Frequency the of the sample rate of wave file. 0-20833
nAvgBytesPerSec	Average data rate. Same as sampling rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of for the data. Always 1
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data. Always 8.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

Olivetti GSM

Added: 01/20/94
Author: Olivetti

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_OLIGSM (0x1000)

wFormatTag	This must be set to WAVE_FORMAT_OLIGSM
nChannels	Number of channels in the wave.(1 for mono), 2
nSamplesPerSec	Frequency the of the sample rate of wave file. 8000
nAvgBytesPerSec	Average data rate. 1633
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data. 196
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data. 2
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. 0

Olivetti ADPCM

Added: 01/20/94
Author: Olivetti

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_OLIADPCM (0x1001)

wFormatTag	This must be set to WAVE_FORMAT_OLIADPCM.
nChannels	Number of channels in the wave. (1, 2)
nSamplesPerSec	Frequency the of the sample rate of wave file. 8000
nAvgBytesPerSec	Average data rate. 4000
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data. 480
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.

wBitsPerSample	This is the number of bits per sample of data. 4
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header. 0

Olivetti CELP

Added: 01/20/94
Author: Olivetti

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_OLISBC (0x1003)

wFormatTag	This must be set to WAVE_FORMAT_OLISBC.
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

Olivetti OPR

Added: 01/20/94
Author: Olivetti

Fact Chunk

This chunk is required for all WAVE formats other than WAVE_FORMAT_PCM. It stores file dependent information about the contents of the WAVE data. It currently specifies the time length of the data in samples.

WAVE Format Header

#define WAVE_FORMAT_OLIOPR (0x1004)

more data not available at time of printing.

wFormatTag	This must be set to WAVE_FORMAT_OLIOPR.
nChannels	Number of channels in the wave.(1 for mono)
nSamplesPerSec	Frequency the of the sample rate of wave file.
nAvgBytesPerSec	Average data rate.
	Playback software can estimate the buffer size using the <**nAvgBytesPerSec**> value.
nBlockAlign	Block Alignment of the data.
	Playback software needs to process a multiple of <**nBlockAlign**> bytes of data at a time, so that the value of <**nBlockAlign**> can be used for buffer alignment.
wBitsPerSample	This is the number of bits per sample of data.
cbSize	The size in bytes of the extra information in the extended `WAVE 'fmt'` header.

RIFF Clipboard Formats

CF_RIFF

Windows 3.1 defines a new clipboard format, CF_RIFF, that allows any RIFF form to be encoded into the clipboard.

CF_WAVE

Windows 3.1 defines a new clipboard format, CF_WAVE, that allows any RIFF form of type WAVE to be encoded into the clipboard.

Registered Clipboard Formats

Because the only way to tell the form of RIFF clipboard data is to read it, an application cannot know if it wants to read the CF_RIFF format or not without getting the data and parsing it. Usually it just wants to look at the form type to determine if it is interested in the data that it contained in the clipboard.

In addition, encoding multiple forms involves a complicated compound RIFF file.

To overcome these problems, Microsoft has defined a standard way to register RIFF clipboard formats. The application should call the Windows API *RegisterClipboardFormat* with a string that specifies the RIFF form of the type that the application is interested. The string should be constructed as follows:

RIFF <FORM>[[' '| u | l][' '| u | l][' '| u | l][' '| u | l]]

where <**form**> is the FOURCC of the form, including spaces. The registration is case insensitive, so form types that have different cases must be uniquely registered. This is accomplished by adding designations of the case of the FOURCC when the <**form**> is not all upper-case.

If any of the characters in the <**form**> are lower-case, then the entire <**form**> must be represented by case designations. Case is designated by appending four characters that represent the case of each character in the <**form**>. The designations are 'u' for uppercase, 'l' for lower-case, and ' ' for space. All non-alphabetics should be represented as spaces.

For example, the form 'Isp ' would be registered as "RIFF Isp ull ". The first character is upper case and therefore the designation character is 'u'. The next two characters are lower-case and therefore the designation characters are both 'l'. The last character is a non-alpha and the designation is therefore a space. As another example, 'L245' would be registered as "RIFF L245 U   "

The CF_RIFF and CF_WAVE formats should still be created in the clipboard in addition to any registered clipboard formats.

Encoding Language of Text

Country Codes

Use one of the following country codes in the **wCountryCode** field:

Country Code	Country
000	None (ignore this field)
001	USA
002	Canada
003	Latin America
030	Greece
031	Netherlands
032	Belgium
033	France
034	Spain
039	Italy
041	Switzerland
043	Austria
044	United Kingdom
045	Denmark
046	Sweden
047	Norway
049	West Germany
052	Mexico
055	Brazil
061	Australia
064	New Zealand
081	Japan
082	Korea
086	People’s Republic of China
088	Taiwan
090	Turkey
351	Portugal
352	Luxembourg
354	Iceland
358	Finland

Language and Dialect Codes

Specify one of the following pairs of language-code and dialect-code values in the **wLanguage** and **wDialect** fields:

Language Code	Dialect Code	Language
0	0	None (ignore these fields)
1	1	Arabic
2	1	Bulgarian
3	1	Catalan
4	1	Traditional Chinese
4	2	Simplified Chinese
5	1	Czech
6	1	Danish
7	1	German
7	2	Swiss German
8	1	Greek
9	1	US English
9	2	UK English
10	1	Spanish
10	2	Spanish Mexican
11	1	Finnish
12	1	French
12	2	Belgian French
12	3	Canadian French
12	4	Swiss French
13	1	Hebrew
14	1	Hungarian
15	1	Icelandic
16	1	Italian
16	2	Swiss Italian
17	1	Japanese
18	1	Korean
19	1	Dutch
19	2	Belgian Dutch
20	1	Norwegian - Bokmal
20	2	Norwegian - Nynorsk
21	1	Polish
22	1	Brazilian Portuguese
22	2	Portuguese
23	1	Rhaeto-Romanic
24	1	Romanian
25	1	Russian
26	1	Serbo-Croatian (Latin)
26	2	Serbo-Croatian (Cyrillic)
27	1	Slovak
28	1	Albanian
29	1	Swedish
30	1	Thai
31	1	Turkish
32	1	Urdu
33	1	Bahasa