Audio Formats Reference

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April 3, 2011

Contents

1 Introduction

This book is intended as a reference for anyone who's ever looked at their collection of audio files and wondered how they worked. Though still a work-in-progress, my goal is to create documentation on the full decoding/encoding process of as many audio formats as possible.

Though to be honest, the audience for this is myself. I enjoy figuring out the little details of how these formats operate. And as I figure them out and implement them in Python Audio Tools, I then add some documentation here on what I've just discovered. That way, when I have to come back to something six months from now, I can return to some written documentation instead of having to go directly to my source code.

Therefore, I try to make my documentation as clear and concise as possible. Otherwise, what's the advantage over simply diving back into the source? Yet this process often turns into a challenge of its own; I'll discover that a topic I thought I'd understood wasn't so easy to grasp once I had to simplify and explain it to some hypothetical future self. Thus, I'll have to learn it better in order to explain it better.

That said, there's still much work left to do. Because it's a repository of my knowledge, it also illustrates the limits of my knowledge. Many formats are little more than "stubs", containing just enough information to extract such metadata as sample rate or bits-per-sample. These are formats in which my Python Audio Tools passes the encoding/decoding task to a binary "black-box" executable since I haven't yet taken the time to learn how to perform that work myself. But my hope is that as I learn more, this work will become more fleshed-out and widely useful.

In the meantime, by including it with Python Audio Tools, my hope is that someone else with some passing interest might also get some use out of what I've learned. And though I strive for accuracy (for my own sake, at least) I cannot guarantee it. When in doubt, consult the references on page 229 for links to external sources which may have additional information.

2 the Basics

2.1 Hexadecimal

In order to understand hexadecimal, it's important to re-familiarize oneself with decimal, which everyone reading this should be familiar with. In ordinary decimal numbers, there are a total of ten characters per digit: 0, 1, 2, 3, 4, 5, 6, 7, 8 and 9. Because there are ten, we'll call it base-10. So the number 675 is made up of the digits 6, 7 and 5 and can be calculated in the following way:

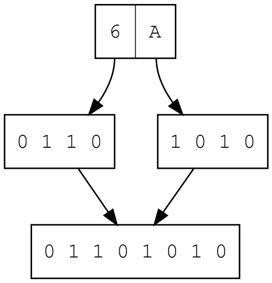
$$(6 \times 10^2) + (7 \times 10^1) + (5 \times 10^0) = 675$$

In hexadecimal, there are sixteen characters per digit: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E and F. A, B, C, D, E and F correspond to the decimal numbers 10, 11, 12, 13, 14 and 15, respectively. Because there are sixteen, we'll call it base-16. So the number 2A3 is made up of the digits 2, A and 3 and can be calculated in the following way:

$$(2 \times 16^2) + (10 \times 16^1) + (3 \times 16^0) = 675$$

Why use hexadecimal? The reason brings us back to binary file formats, which are made up of bytes. Each byte is made up of 8 bits and can have a value from 0 to 255, in decimal. Representing a binary file in hexadecimal means a byte requires exactly two digits with values from 0 to FF. That saves us a lot of space versus trying to represent bytes in decimal.

Hexadecimal has another important property when dealing with binary data. Because each digit has 16 possible values, each hexadecimal digit represents exactly 4 bits ($16 = 2^4$). This makes it very easy to go back and forth between hexadecimal and binary. For instance, let's take the byte 6A:



Hex	Binary	Decimal	Hex	Binary	Decimal
0	0 0 0 0	0	8	1 0 0 0	8
1	0 0 0 1	1	9	1 0 0 1	9
2	0 0 1 0	2	A	1 0 1 0	10
3	0 0 1 1	3	B	1 0 1 1	11
4	0 1 0 0	4	C	1 1 0 0	12
5	0 1 0 1	5	D	1 1 0 1	13
6	0 1 1 0	6	E	1 1 1 0	14
7	0 1 1 1	7	F	1 1 1 1	15

Going from binary to hexadecimal is a simple matter of reversing the process.

2.2 Signed Integers

Signed integers are typically stored as “2’s-complement” values. To decode them, one needs to know the integer’s size in bits, its topmost (most-significant) bit value and the value of its remaining bits.

$$\text{signed value} = \begin{cases} \text{remaining bits} & \text{if topmost bit} = 0 \\ \text{remaining bits} - (2^{\text{integer size}-1}) & \text{if topmost bit} = 1 \end{cases} \tag{2.1}$$

For example, take an 8-bit integer whose bit values are `00000101`. Since the topmost bit is `0`, its value is simply `0000101`, which is 5 in base-10 ($2^2 + 2^0 = 5$).

Next, let’s take an integer whose bit values are `11111011`. Its topmost bit is `1` and its remaining bits are `1111011`, which is 123 in base-10 ($2^6 + 2^5 + 2^4 + 2^3 + 2^1 + 2^0 = 123$). Therefore:

$$\begin{aligned} \text{signed value} &= 123 - 2^{8-1} \\ &= 123 - 128 \\ &= -5 \end{aligned}$$

Transforming a signed integer into its unsigned 2’s-complement value is a simple matter of reversing the process.

$$\text{unsigned value} = \begin{cases} \text{signed value} & \text{if signed value} \geq 0 \\ 2^{\text{integer size}} - (-\text{signed value}) & \text{if signed value} < 0 \end{cases} \tag{2.2}$$

For example, let’s convert the value -20 to a signed, 8-bit integer:

$$\begin{aligned} \text{unsigned value} &= 2^8 - (- - 20) \\ &= 256 - 20 \\ &= 236 \end{aligned}$$

which is `11101100` in binary ($2^7 + 2^6 + 2^5 + 2^3 + 2^2 = 236$).

2.3 Endianness

You will need to know about endianness anytime a single value spans multiple bytes. As an example, let's take the first 16 bytes of a small RIFF WAVE file:

```
52 49 46 46 54 9b 12 00  57 41 56 45 66 6d 74 20
```

The first four bytes are the ASCII string 'RIFF' (`0x52 0x49 0x46 0x46`). The next four bytes are a 32-bit unsigned integer which is a size value. Reading from left to right, that value would be `0x549B1200`. That's almost 1.5 gigabytes. Since this file is nowhere near that large, we're clearly not reading those bytes correctly.

The key is that RIFF WAVE files are 'little endian'. In plain English, that means we have to read in those bytes from right to left. Thus, the value is actually `0x00129B54`. That's a little over 1 megabyte, which is closer to our expectations.

Remember that little endian reverses the bytes, not the hexadecimal digits. Simply reversing the string to `0x0021B945` is not correct.

When converting a signed, little-endian value to an integer, the 2's-complement decoding comes *after* one performs the endianness reversing. For example, given a signed 16-bit little-endian value of `0xFBFF`, one firsts reorders the bytes to `0xFFFB` before decoding it to a signed value ($32763 - 2^{15} = -5$).

Conversely, when converting a signed integer to a little-endian value, the endian reversing comes *after* one performs the 2's-complement encoding.

This is covered in more depth in the bitstreams section on page 17.

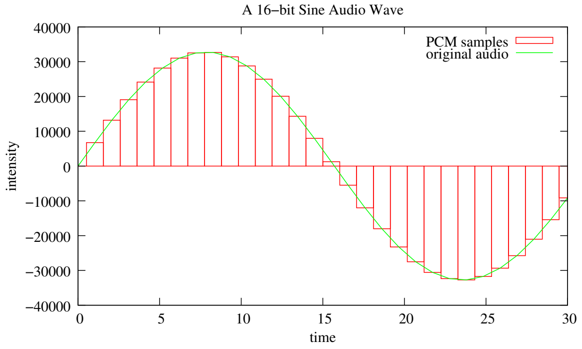
2.4 Character Encodings

Many audio formats store metadata, which contains information about the song's name, artist, album and so forth. This information is stored as text, but it's important to know what sort of text in order to read it and display it properly.

As an example, take the simple character `é`. In Latin-1 encoding, it is stored as a single byte `0xE9`. In UTF-8 encoding, it is stored as the bytes `0xC3A9`. In UTF-16BE encoding, it is stored as the bytes `0x00E9`.

Although decoding and encoding text is a complex subject beyond the scope of this document, you must always be aware that metadata may not be 7-bit ASCII text and should handle it properly in whatever encoding is supported by the metadata formats. Look to your programming tools for libraries to assist in Unicode text handling.

2.5 PCM



Pulse-Code Modulation is a method for transforming an analog audio signal into a digital representation. It takes that signal, ‘samples’ its intensity at discrete intervals and yields a stream of signed integer values. By replaying those values to a speaker at the same speed and intensity, a close approximation of the original signal is produced.

Let’s take some example bytes from a CD-quality PCM stream:

```
1B 00 43 FF  1D 00 45 FF  1C 00 4E FF  1E 00 59 FF
```

CD-quality is 16-bit, 2 channel, 44100Hz. 16-bit means those bytes are split into 16-bit signed, little-endian samples. Therefore, our bytes are actually the integer samples:

```
27 -189 29 -187 28 -178 30 -167
```

The number of channels indicates how many speakers the signal supports. 2 channels means the samples are sent to 2 different speakers. PCM interleaves its samples, sending one sample to each channel simultaneously before moving on to the next set. In the case of 2 channels, the first sample is sent to the left speaker and the second is sent to the right speaker. So, our stream of data now looks like:

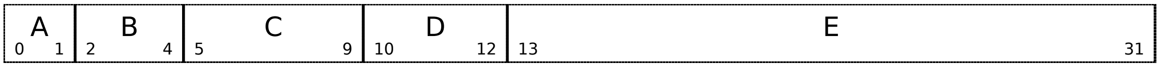
left speaker	right speaker
27	-189
29	-187
28	-178
30	-167

44100Hz means those pairs of samples are sent at the rate of 44100 per second. Thus, our set of 4 samples takes precisely 1/11025th of a second when replayed.

A channel-independent block of samples is commonly referred to as a ‘frame’. In this example, we have a total of 4 PCM frames. However, the term ‘frame’ appears a lot in digital audio. It is important not to confuse a PCM frame with a CD frame (a block of audio 1/75th of a second long), an MP3 frame, a FLAC frame or any other sort of frame.

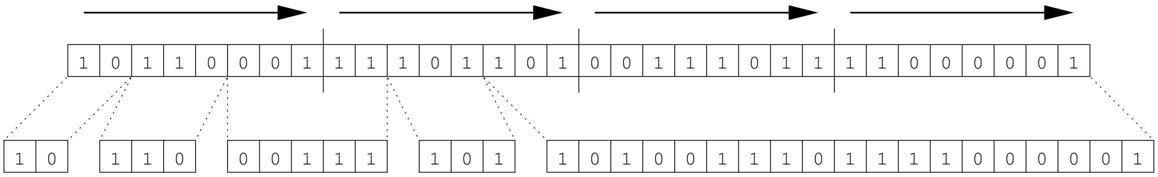
2.6 Bitstreams

Many formats are broken up into pieces smaller than an individual byte. As an example, let's take a 32-bit structure of data broken up into 5 fields as follows:



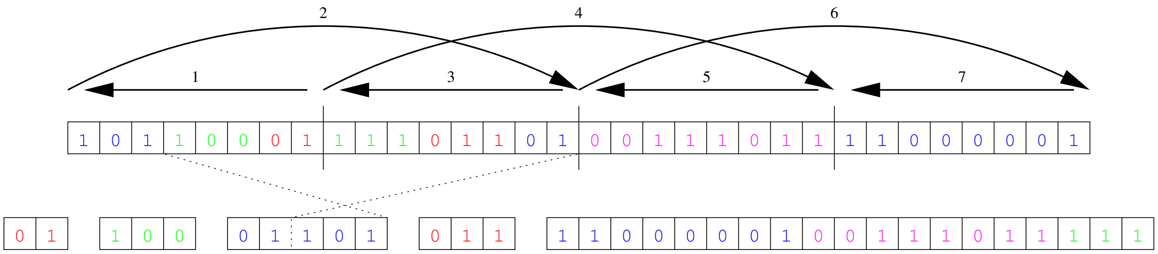
Note that field A is 2 bits, B is 3 bits, C is 5 bits, D is 3 bits and E is 19 bits.

Given the bytes `B1 ED 3B C1` in hexadecimal, how we place those bytes within our structure depends whether or not our stream is big-endian or little-endian. For big-endian, the breakdown is largely intuitive:



field	base-2	base-16	base-10
A	`10`	`2`	`2`
B	`110`	`6`	`6`
C	`0 0111`	`07`	`7`
D	`101`	`5`	`5`
E	`101 0011 1011 1100 0001`	`53BC1`	`342977`

However, for little-endian streams, the same bits are read quite differently:

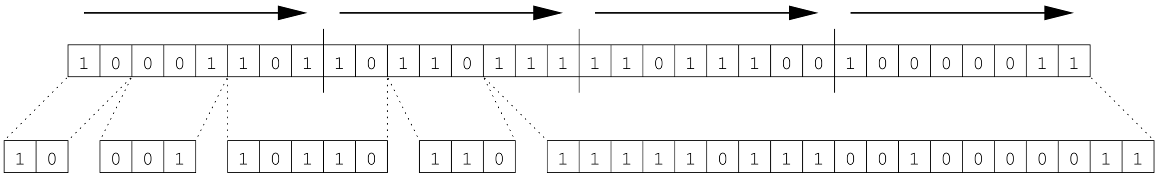


field	base-2	base-16	base-10
A	`01`	`1`	`1`
B	`100`	`4`	`4`
C	`0 1101`	`0D`	`13`
D	`011`	`3`	`3`
E	`110 0000 1001 1101 1111`	`609DF`	`395743`

This arrangement seems bizarre and counterintuitive when using the same hexadecimal-to-binary conversion we've been using thus far. However, if we *reverse* our conversion process and place the least-significant bits on the left, as follows:

Hex	Binary	Decimal	Hex	Binary	Decimal
0	0 0 0 0	0	8	0 0 0 1	8
1	1 0 0 0	1	9	1 0 0 1	9
2	0 1 0 0	2	A	0 1 0 1	10
3	1 1 0 0	3	B	1 1 0 1	11
4	0 0 1 0	4	C	0 0 1 1	12
5	1 0 1 0	5	D	1 0 1 1	13
6	0 1 1 0	6	E	0 1 1 1	14
7	1 1 1 0	7	F	1 1 1 1	15

The bytes `B1 ED 3B C1` are then laid out differently also and can now be read in a linear fashion:



As a further example of how this works, we'll convert the first four fields' binary digits to base-10:

$$\begin{aligned}
\text{A} &= (1 \times 2^0) + (0 \times 2^1) = 1 \\
\text{B} &= (0 \times 2^0) + (0 \times 2^1) + (1 \times 2^2) = 4 \\
\text{C} &= (1 \times 2^0) + (0 \times 2^1) + (1 \times 2^2) + (1 \times 2^3) + (0 \times 2^4) = 13 \\
\text{D} &= (1 \times 2^0) + (1 \times 2^1) + (0 \times 2^2) = 3
\end{aligned}$$

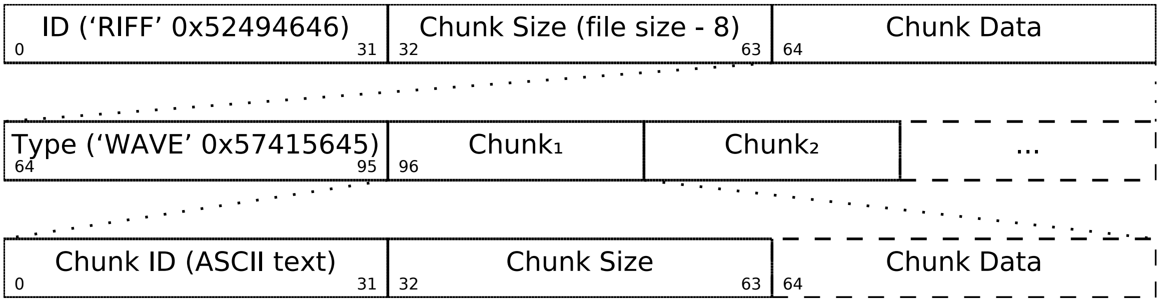
Computing field E is left as an exercise for the reader.

Naturally, these fields from the same bytes have the same values as before; only our way of visualizing them has changed.

3 Waveform Audio File Format

The Waveform Audio File Format is the most common form of PCM container. What that means is that the file is mostly PCM data with a small amount of header data to tell applications what format the PCM data is in. Since RIFF WAVE originated on Intel processors, everything in it is little-endian.

3.1 the RIFF WAVE Stream



‘Chunk Size’ is the total size of the chunk, minus 8 bytes for the chunk header.

3.2 the Classic ‘fmt’ Chunk

Wave files with 2 channels or less, and 16 bits-per-sample or less, use a classic ‘fmt’ chunk to indicate its PCM data format. This chunk is required to appear before the ‘data’ chunk.

Chunk ID (‘fmt ’ 0x666D7420) 0 31	Chunk Size (16) 32 63
Compression Code (0x0001) 64 79	Channel Count 80 95
Sample Rate 96 127	
Average Bytes per Second 128 159	
Block Align 160 175	Bits per Sample 176 191

$$\text{Average Bytes per Second} = \frac{\text{Sample Rate} \times \text{Channel Count} \times \text{Bits per Sample}}{8} \tag{3.1}$$

$$\text{Block Align} = \frac{\text{Channel Count} \times \text{Bits per Sample}}{8} \tag{3.2}$$

3.3 the WAVEFORMATEXTENSIBLE ‘fmt’ Chunk

Wave files with more than 2 channels or more than 16 bits-per-sample should use a WAVEFORMATEXTENSIBLE ‘fmt’ chunk which contains additional fields for channel assignment.

<table>
<tr><td colspan="2">Chunk ID (‘fmt ’ 0x666D7420)
0 31</td><td colspan="2">Chunk Size (40)
32 63</td></tr>
<tr><td colspan="2">Compression Code (0xFFFE)
64 79</td><td colspan="2">Channel Count
80 95</td></tr>
<tr><td colspan="4">Sample Rate
96 127</td></tr>
<tr><td colspan="4">Average Bytes per Second
128 159</td></tr>
<tr><td colspan="2">Block Align
160 175</td><td colspan="2">Bits per Sample
176 191</td></tr>
<tr><td colspan="2">CB Size (22)
192 207</td><td colspan="2">Valid Bits per Sample
208 223</td></tr>
<tr><td>Front Right of Center
224</td><td>Front Left of Center
225</td><td>Rear Right
226</td><td>Rear Left
227</td></tr>
<tr><td>LFE
228</td><td>Front Center
229</td><td>Front Right
230</td><td>Front Left
231</td></tr>
<tr><td>Top Back Left
232</td><td>Top Front Right
233</td><td>Top Front Center
234</td><td>Top Front Left
235</td></tr>
<tr><td>Top Center
236</td><td>Side Right
237</td><td>Side Left
238</td><td>Back Center
239</td></tr>
<tr><td>Undefined
240 245</td><td>Top Back Right
246</td><td>Top Back Center
247</td><td>Undefined
248 255</td></tr>
<tr><td colspan="4">Sub Format (0x01000000000001000800000aa00389b71)
256 383</td></tr>
</table>

Note that the ‘Average Bytes per Second’ and ‘Block Align’ fields are calculated the same as a classic fmt chunk.

3.4 the ‘data’ Chunk

<table>
<tr><td>Chunk ID (‘data’ 0x64617461)
0 31</td><td>Chunk Size
32 63</td></tr>
<tr><td colspan="2">PCM Data
64</td></tr>
</table>

‘PCM Data’ is a stream of PCM samples stored in little-endian format.

3.5 Channel Assignment

Channels whose bits are set in the WAVEFORMATEXTENSIBLE ‘fmt’ chunk appear in the following order:

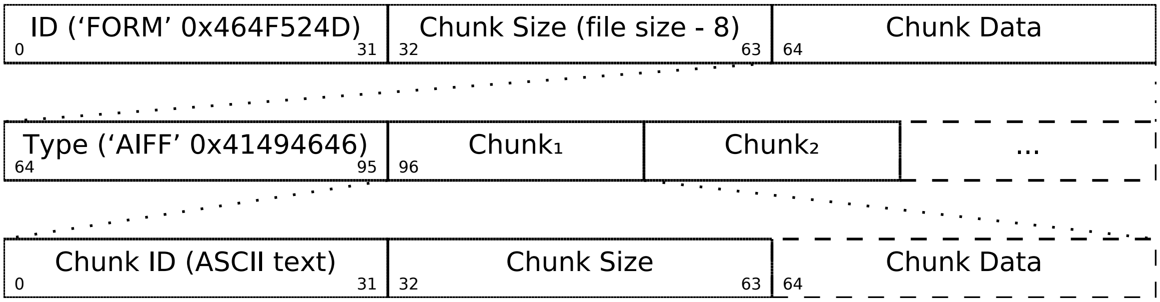
Index	Channel	Mask Bit
1	Front Left	`0x1`
2	Front Right	`0x2`
3	Front Center	`0x4`
4	LFE	`0x8`
5	Back Left	`0x10`
6	Back Right	`0x20`
7	Front Left of Center	`0x40`
8	Front Right of Center	`0x80`
9	Back Center	`0x100`
10	Side Left	`0x200`
11	Side Right	`0x400`
12	Top Center	`0x800`
13	Top Front Left	`0x1000`
14	Top Front Center	`0x2000`
15	Top Front Right	`0x4000`
16	Top Back Left	`0x8000`
17	Top Back Center	`0x10000`
18	Top Back Right	`0x20000`

For example, if the file’s channel mask is set to `0x33`, it contains the channels ‘Front Left’, ‘Front Right’, ‘Back Left’ and ‘Back Right’, in that order.

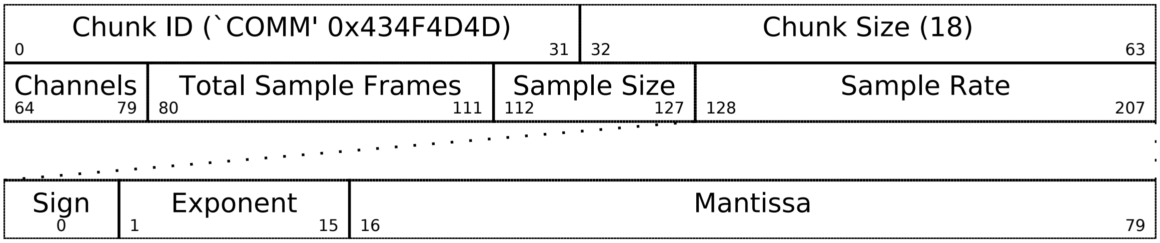
4 Audio Interchange File Format

AIFF is the Audio Interchange File Format. It is popular on Apple computers and is a precursor to the more widespread WAVE format. All values in AIFF are stored as big-endian.

4.1 the AIFF File Stream



4.2 the COMM Chunk



The 'Sample Rate' field is an 80-bit IEEE Standard 754 floating point value instead of the big-endian integers common to all the other fields.

$$\text{Value} = (-)\frac{\text{Mantissa}}{2^{63}} \times 2^{\text{Exponent}-16383}$$

For example, given a sign bit of 0, an exponent value of 0x400E and a mantissa value of 0xAC44000000000000:

$$\begin{aligned}\text{Value} &= \frac{12413046472939929600}{2^{63}} \times 2^{16398-16383} \\ &= 1.3458251953125 \times 2^{15} \\ &= 44100.0\end{aligned}$$

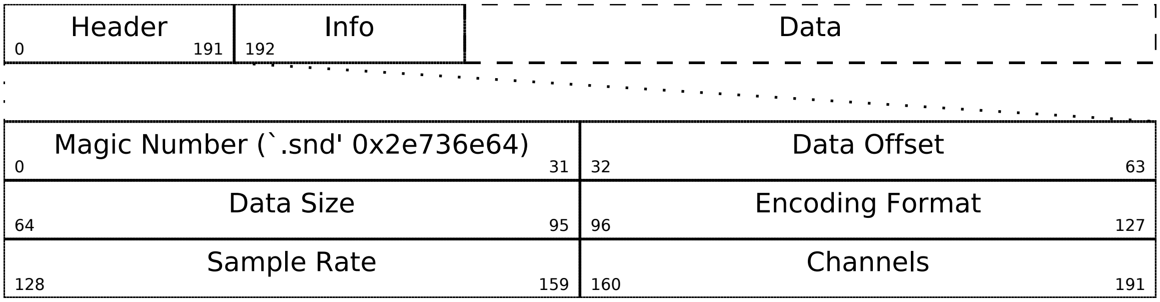
4.3 the SSND Chunk

Chunk ID (`SSND' 0x53534E44) 0 31	Chunk Size 32 63
Offset 64 95	Block Size 96 127
PCM Data 128	

5 Sun AU

The AU file format was invented by Sun Microsystems and also used on NeXT systems. All values in AU are stored as big-endian. It supports a wide array of data formats, including µ-law logarithmic encoding, but can also be used as a PCM container.

5.1 the Sun AU File Stream



value	encoding format
1	8-bit G.711 µ-law
2	8-bit linear PCM
3	16-bit linear PCM
4	24-bit linear PCM
5	32-bit linear PCM
6	32-bit IEEE floating point
7	64-bit IEEE floating point
8	Fragmented sample data
9	DSP program
10	8-bit fixed point
11	16-bit fixed point
12	24-bit fixed point
13	32-bit fixed point
18	16-bit linear with emphasis
19	16-bit linear compressed
20	16-bit linear with emphasis and compression
21	Music kit DSP commands
23	4-bit ISDN µ-law compressed using the ITU-T G.721 ADPCM voice data encoding scheme
24	ITU-T G.722 ADPCM
25	ITU-T G.723 3-bit ADPCM
26	ITU-T G.723 5-bit ADPCM
27	8-bit G.711 A-law

6 Shorten

Shorten is one of the earliest lossless audio compression formats. Though superseded by FLAC and other formats, it remains interesting from a historical perspective.

6.1 Shorten Data Types

Notably, almost nothing in the Shorten file format is byte-aligned. Instead, it uses its own set of variable-length types which I'll refer to as `unsigned`, `signed` and `long`.

An `unsigned` field of a certain 'size' means we first take a unary-encoded[1], number of high bits and combine the resulting value with 'size' number of low bits. For example, given a 'size' of 2 and the bits '`0 0 1 1 1`', the high unary value of '`0 0 1`' combines with the low raw value of '`1 1`' resulting in a decimal value of 11.

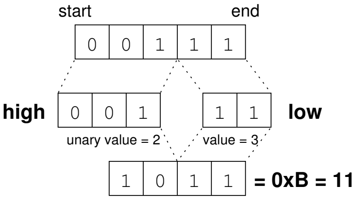


Figure 6.1: Unsigned

A `signed` field is similar, but its low value contains one additional trailing bit for the sign value.

$$\text{signed value} = \begin{cases} \text{unsigned value} & \text{if sign bit} = 0 \\ -\text{unsigned value} - 1 & \text{if sign bit} = 1 \end{cases}$$

For example, given a 'size' of 3 and the bits '`0 1 1 0 1 1`', the high unary value of '`0 1`' combines with the low raw value of '`1 0 1`' and the sign bit '`1`' resulting in a decimal value of -14. Note that the sign bit is counted separately, so we're actually reading 4 additional bits after the unary value in this case.

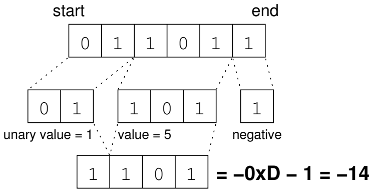


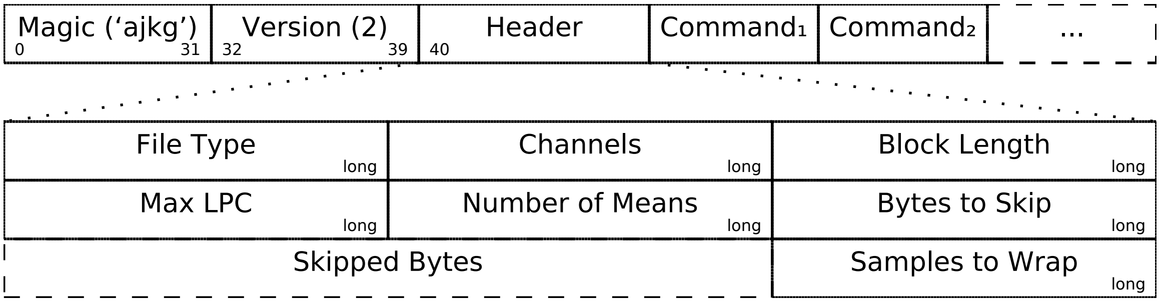
Figure 6.2: Signed

Lastly, and most confusingly, a `long` field is the combination of two separate `unsigned` fields. The first, of size 2, determines the size value of the second. For example, given the bits '`1 1 1 1 1 0 1`', the first `unsigned` field of '`1 1 1`' has the value of 3 (unary 0 combined with a raw value of 3) - which is the size of the next `unsigned` field. That field, in turn, consists of the bits '`1 1 0 1`' which is 5 (unary 0 combined with a raw value of 5). So, the value of the entire `long` field is 5.

A Shorten file consists almost entirely of these three types in various sizes. Therefore, when one reads "`unsigned(3)`" in a Shorten field description, it means an `unsigned` field of size 3.

[1] In this instance, unary-encoding is a simple matter of counting the number of 0 bits before the next 1 bit. The resulting sum is the value.

6.2 the Shorten File Stream



file type	format
0	lossless µ-Law
1	signed 8 bit
2	unsigned 8 bit
3	signed 16 bit, big-endian
4	unsigned 16 bit, big-endian
5	signed 16 bit, little-endian
6	unsigned 16 bit, little-endian
7	lossy µ-Law
8	new µ-Law with zero mapping
9	lossless a-Law
10	lossy a-Law
11	Microsoft .wav
12	Apple .aiff

'Channels' is the number of channels in the audio stream. 'Block Length' is the length of each command block, in samples. 'Max LPC' is the maximum LPC value a block may have. 'Samples to Wrap' is the number of samples to be wrapped around from the top of an output block to the bottom. This will be explained in more detail in the decoding section.

6.3 Shorten Decoding

Internally, a Shorten file acts as a list of commands to be executed by a tiny virtual machine.[2] Each command is a `unsigned(2)` field followed by zero or more arguments.

value	command	value	command
0	`DIFF0`	5	`BLOCKSIZE`
1	`DIFF1`	6	`BITSHIFT`
2	`DIFF2`	7	`QLPC`
3	`DIFF3`	8	`ZERO`
4	`QUIT`	9	`VERBATIM`

6.3.1 the DIFF Command

All four `DIFF` commands are structured the same:

Energy Size unsigned(3)	Residual₁ signed(energy)	Residual₂ signed(energy)	...	Residualₓ signed(energy)

There are 'Block Size' number of residuals per `DIFF` (whose initial value is determined by the Shorten header) and each one's size is determined by 'Energy Size'. The process of transforming these residuals into samples depends on the `DIFF` command and the values of previously decoded samples.

Command	Calculation
`DIFF0`	$Sample_i = Residual_i + Coffset^a$
`DIFF1`	$Sample_i = Sample_{i-1} + Residual_i$
`DIFF2`	$Sample_i = (2 \times Sample_{i-1}) - Sample_{i-2} + Residual_i$
`DIFF3`	$Sample_i = (3 \times (Sample_{i-1} - Sample_{i-2})) + Sample_{i-3} + Residual_i$

[a] See page 32

For example, given a `DIFF1` command at the stream's beginning and the residual values 10, 1, 2, -2, 1 and -1, samples are calculated as follows:

Index	Residual	Sample
-1	(before stream)	(not output) **0**
0	10	$0 + 10 = \mathbf{10}$
1	1	$10 + 1 = \mathbf{11}$
2	2	$11 + 2 = \mathbf{13}$
3	-2	$13 - 2 = \mathbf{11}$
4	1	$11 + 1 = \mathbf{12}$
5	-1	$12 - 1 = \mathbf{11}$

[2] Interestingly, although Shorten's successor, FLAC, presents its input as frames and subframes, references to a FLAC virtual machine are still present in its source code.

6.3.2 Channels and Wrapping

The audio commands `DIFF`, `QLPC` and `ZERO` send their samples to channels in order. For example, a stream of `DIFF` commands in a 2 channel stereo stream (a very typical configuration) sends $\texttt{DIFF}_1$ to the left channel, $\texttt{DIFF}_2$ to the right channel, $\texttt{DIFF}_3$ to left channel, $\texttt{DIFF}_4$ to the right channel and so on.

However, recall that most of the `DIFF` commands require previously decoded samples as part of their calculation. What this means is that $\texttt{DIFF}_3$ takes the last few samples from $\texttt{DIFF}_1$ in order to apply its residuals (since both are on the left channel) and $\texttt{DIFF}_4$ takes the last few samples from $\texttt{DIFF}_2$.

This is where the header's 'Samples to Wrap' field comes into play. Its value is the number of samples to be wrapped from the top of the buffer to its pre-zero values. For example, if 'Sample Count' is 256 and 'Samples to Wrap' is 3 (another typical configuration), Buffer_{-1} takes the value of Buffer_{255}, Buffer_{-2} takes the value of Buffer_{254}, and Buffer_{-3} takes the value of Buffer_{253}. However, these pre-zero starting-point values are obviously not re-output when the buffer is finally completed and returned.

6.3.3 the QUIT Command

This command takes no arguments. It indicates the Shorten stream is finished and decoding is completed.

6.3.4 the BLOCKSIZE Command

This command takes a single `long` argument whose value is the new 'Block Size'. In effect, it modifies that variable in the Shorten virtual machine.

6.3.5 the ZERO Command

This command takes no arguments. It simply generates 'Block Size' number of zero samples into the current channel's output buffer.

6.3.6 the BITSHIFT Command

This commands takes a single `unsigned(2)` value and modifies the 'bitshift' variable in the Shorten virtual machine. This value is how many bits to left-shift all output samples prior to returning them.

For example, imagine a scenario in which all the samples in a set of blocks have 0 for their rightmost (least significant) bit. Setting a bitshift of 1 allows us to ignore that bit during calculation which, in turn, allows us to store those samples more efficiently.

Note that bit shifting is applied *after* channel wrapping and 'coffset' calculation[3].

[3] See page 32

6.3.7 the QLPC Command

The QLPC command is structured as follows:

Energy Size unsigned(2)	LPC Count unsigned(2)	LPC Coeff1 signed(5)	...	LPC Coeffx signed(5)
Residual1 signed(energy)	Residual2 signed(energy)	Residual3 signed(energy)	...	Residualx signed(energy)

So, given a set of LPC coefficients and a set of residuals, samples are calculated using the following formula:

$$\text{Sample}_i = \left\lfloor \frac{2^5 + \sum_{j=0}^{Count-1} \text{LPC Coefficient}_j \times \text{Sample}_{i-j-1}}{2^5} \right\rfloor + \text{Residual}_i \qquad (6.1)$$

This simply means we're taking the sum of the calculated values from 0 to LPC Count - 1, bit-shifting that sum down and added the residual when determining the current sample. As with the DIFF commands, previously encoded samples (possibly from previous commands) are used to calculate the current sample.

For example, given the LPC Coefficients and previously encoded samples:

LPC Coefficient$_0$	21	Sample$_1$	-2
LPC Coefficient$_1$	2	Sample$_2$	-3
LPC Coefficient$_2$	7	Sample$_3$	-2

Index	Residual	Sample
1		$-\mathbf{2}$
2		$-\mathbf{3}$
3		$-\mathbf{2}$
4	1	$\left\lfloor \frac{2^5+(21\times-2)+(2\times-3)+(7\times-2)}{2^5} \right\rfloor + 1 = \lfloor\frac{32-62}{32}\rfloor + 1 = -1 + 1 = \mathbf{0}$
5	-2	$\left\lfloor \frac{2^5+(21\times\mathbf{0})+(2\times-2)+(7\times-3)}{2^5} \right\rfloor - 2 = \lfloor\frac{32-25}{32}\rfloor - 2 = 0 - 2 = -\mathbf{2}$
6	-1	$\left\lfloor \frac{2^5+(21\times\mathbf{-2})+(2\times\mathbf{0})+(7\times-2)}{2^5} \right\rfloor - 1 = \lfloor\frac{32-56}{32}\rfloor - 1 = -1 - 1 = -\mathbf{2}$

Unfortunately, there's one more wrinkle to consider for proper QLPC command decoding: the 'coffset'. How to calculate this value will be covered in the next section. But when a QLPC command is encountered, the coffset value is subtracted from the QLPC's warm-up samples (taken from the top of the previous command, for the current channel). Then that coffset value is re-added to our output samples after calculation.

For example, given a 'coffset' value of 5, one would subtract 5 from Sample$_{-3}$, Sample$_{-2}$ and Sample$_{-1}$, perform the QLPC calculation and then add 5 to our Sample$_0$, Sample$_1$, Sample$_2$, ... , Sample$_{255}$ before returning those values.

6.3.8 the Coffset

Calculating the 'coffset' value for a given command on a given channel requires a set of 'offset' values (whose count equals the 'Number of Means', from the Shorten header) and the 'Number of Means' value itself.

$$\text{coffset} = \frac{\frac{\text{nmeans}}{2} + \sum_{i=0}^{\text{nmeans}-1} \text{offset}_i}{\text{nmeans}} \tag{6.2}$$

For example, given a 'Number of Means' value of 4 and offsets of:

offset_0	32
offset_1	28
offset_2	17
offset_3	14

$$\text{coffset} = \frac{\frac{4}{2} + (32 + 28 + 17 + 14)}{4} = \frac{93}{4} = \mathbf{23} \tag{6.3}$$

The next obvious question is where to those 'offset' values come from? They're actually a queue of (mostly) sample value averages on the given channel. So once we've decoded offset for command_5 on channel 0, offset_0 takes the value of offset_1, offset_1 takes the value of offset_2, offset_2 takes the value of offset_3, and command_5's offset becomes the new offset_3.

However, the offset is not entirely a sample average. Its actual formula is as follows:

$$\text{offset} = \frac{\frac{\text{block size}}{2} + \sum_{i=0}^{\text{block size}-1} \text{sample}_i}{\text{block size}} \tag{6.4}$$

For example, if a command with a 'block size' of 256 has samples that total 1056, its offset value is:

$$\text{offset} = \frac{\frac{256}{2} + 1056}{256} = \frac{1184}{256} = \mathbf{4} \tag{6.5}$$

6.3.9 the VERBATIM Command

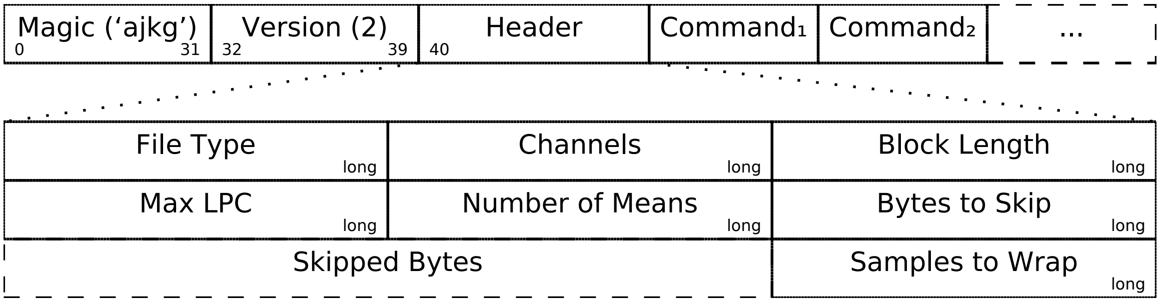
This command is for generating raw, non-audio data such as .wav or .aiff chunks and is structured as follows:

Total Bytes unsigned(5)	Byte₁ unsigned(8)	Byte₂ unsigned(8)	...	Byteₓ unsigned(8)

These chunks of raw data are expected to be written in the order they appear in the Shorten file.

6.4 Shorten Encoding

For the purposes of Shorten encoding, one needs an entire PCM container file and its PCM values, number of channels and bits per sample. Recall that nearly the entire Shorten file format is made up of three variable-length data types with no byte-alignment of any sort. Because of that, there's no way to "rewind" the Shorten stream and replace values. Therefore, everything encoded to Shorten must be written in order.



Remember that `long` variables contain a size `unsigned(2)` field followed by its value. So what's the best size to use for a given value? Quite frankly, in my opinion, any small size will do. Most of a Shorten file is encoded residuals, so wasting a handful of bytes in the header won't make any difference.

A good set of header values to use are as follows:

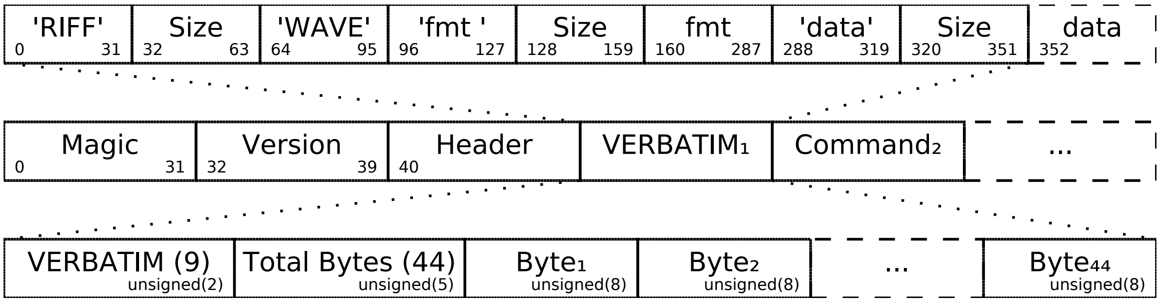
field	value
Field Type	**2** for unsigned 8 bit input **5** for signed, little-endian, 16 bit input **3** for signed, big-endian, 16 bit input
Channels	from input
Block Length	**256**
Max LPC	**0**
Number of Means	**0**
Bytes to Skip	**0**
Samples to Wrap	**3**

The remainder of the stream is various Shorten commands. I'll be limiting this encoding documentation to the `DIFF1`, `DIFF2`, `DIFF3`, `QUIT`, `BLOCKSIZE`, `ZERO` and `VERBATIM` commands. `QLPC`'s implementation is actually broken in the reference decoder[4] and tends not to produce smaller files. And, by omitting `DIFF0` which is rarely used, we can avoid 'coffset' calculation during encoding.

[4] By limiting its accumulator to a 32-bit integer, it's prone to overflow at high LPC counts.

6.4.1 the VERBATIM Command

PCM containers such as Wave and AIFF consist of several blocks of data, one of which contains a large chunk of PCM data. Shorten encodes these files by turning non-audio data at the beginning and end of the container into `VERBATIM` commands, often in the following format:



These `VERBATIM` commands *must* appear in the Shorten stream in the same order as they appear in the PCM container itself.

6.4.2 the BLOCKSIZE Command

Most `DIFF` and `ZERO` commands should contain ‘Block Length’ number of samples, as indicated in the header. But when the end of the input stream is reached and a different number of samples are all that remain, a `BLOCKSIZE` is required. This is simply an `unsigned(2)` variable with a value of 5 (indicating the `BLOCKSIZE` command) followed by a `long` variable containing the new ‘Block Length’.

6.4.3 the QUIT Command

When no more samples remain and all `VERBATIM` commands have been delivered (a Wave file may contain additional chunks after the ‘data’ chunk, for example), the `QUIT` command indicates the end of the Shorten stream. This is an `unsigned(2)` variable with the value 4.

However, the stream *must* be padded such that the total stream length, minus 5 bytes for the magic number and version, is a multiple of 4 bytes. If this padding is not performed, the reference decoder’s bit stream implementation will exit with an error.

6.4.4 the ZERO Command

When a block’s entire set of samples are 0, we can generate a `ZERO` command to represent them, which is simply the `unsigned(2)` command variable of 8.

6.4.5 the DIFF Commands

The bulk of a Shorten file is `DIFF` commands, which are `unsigned(2)` command variables 1, 2, and 3 for `DIFF1`, `DIFF2` and `DIFF3`, respectively, followed by the same argument syntax:

Energy Size	Residual$_1$	Residual$_2$	...	Residual$_x$
unsigned(3)	signed(energy)	signed(energy)		signed(energy)

Given a set of input samples, we decide which `DIFF` command to use by calculating their minimum delta sum, which is best explained by example:

index	sample	Δ^1	Δ^2	Δ^3
-1	0			
0	18	-18		
1	20	-2	-16	
2	26	-6	4	-20
3	24	2	-8	12
4	24	0	2	-10
5	23	1	-1	3
6	21	2	-1	0
7	24	-3	5	-6
8	23	1	-4	9
9	20	3	-2	-2
10	18	2	1	-3
11	18	0	2	-1
12	17	1	-1	3
13	17	0	1	-2
14	20	-3	3	-2
15	23	-3	0	3
16	21	2	-5	5
17	23	-2	4	-9
18	22	1	-3	7
19	18	4	-3	0
	$\|sum\|$	56	66	97

In this example, the $|sum|$ of Δ^1 is the smallest value. Therefore, the best command to use for this set of samples is `DIFF1`. Once we know which `DIFF` command to use for a given set of input samples, calculating the command's set of residual values can be done automatically:

command	calculation
`DIFF1`	$\text{Residual}_i = \text{Sample}_i - \text{Sample}_{i-1}$
`DIFF2`	$\text{Residual}_i = \text{Sample}_i - ((2 * \text{Sample}_{i-1}) - \text{Sample}_{i-2})$
`DIFF3`	$\text{Residual}_i = \text{Sample}_i - ((3 \times (Sample_{i-1} - Sample_{i-2})) + Sample_{i-3})$

In this example, our residuals for `DIFF1` are: 18, 2, 6, -2, 0, -1, -2, 3, -1, -3, -2, 0, -1, 0, 3, 3, -2, 2, -1, -4.

Finally, given a set of residual values, we need to determine their 'Energy Size'. This is done by selecting the smallest value of 'x' such that:

$$\text{sample count} \times 2^x > \sum_{i=0}^{\text{residual count}-1} |\text{residual}_i| \tag{6.6}$$

To finish our example, given we have 20 residuals:

index	residual_i	$\lvert\text{residual}_i\rvert$
0	18	18
1	2	2
2	6	6
3	-2	2
4	0	0
5	-1	1
6	-2	2
7	3	3
8	-1	1
9	-3	3
10	-2	2
11	0	0
12	-1	1
13	0	0
14	3	3
15	3	3
16	-2	2
17	2	2
18	-1	1
19	-4	4
	$\lvert sum\rvert$	56

$$20 \times 2^0 \leq 56$$
$$20 \times 2^1 \leq 56$$
$$20 \times 2^2 > 56$$

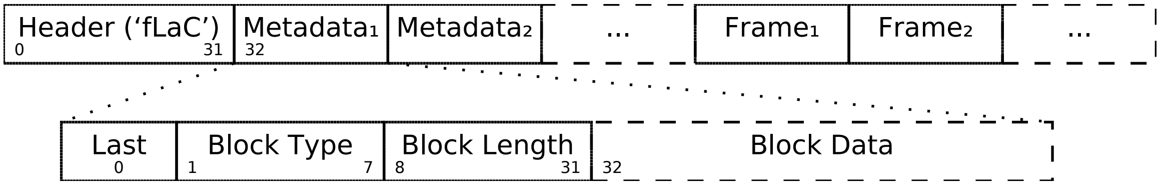
Therefore, the best 'Energy Size' for this set of residuals is 2.

7 Free Lossless Audio Codec

FLAC compresses PCM audio data losslessly using predictors and a residual. FLACs contain checksumming to verify their integrity, contain comment tags for metadata and are streamable.

Except for the contents of the VORBIS_COMMENT metadata block, everything in FLAC is big-endian.

7.1 the FLAC File Stream

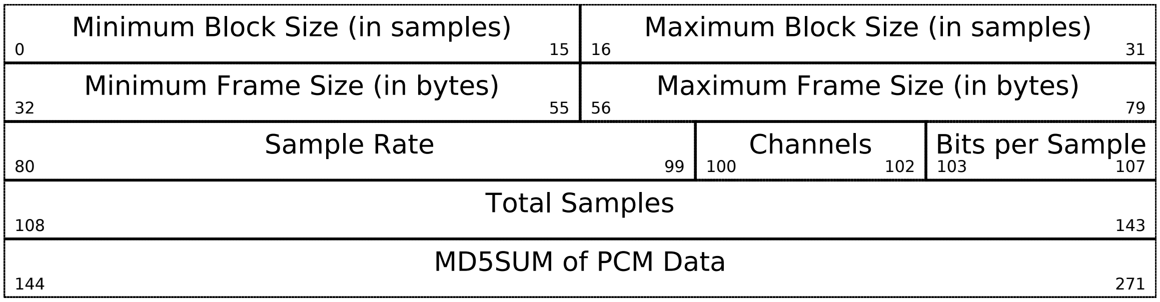


‘Last’ is 0 when there are no additional metadata blocks and 1 when it is the final block before the the audio frames. ‘Block Length’ is the size of the metadata block data to follow, not including the header.

Block Type	Block
0	STREAMINFO
1	PADDING
2	APPLICATION
3	SEEKTABLE
4	VORBIS_COMMENT
5	CUESHEET
6	PICTURE
7-126	reserved
127	invalid

7.2 FLAC Metadata Blocks

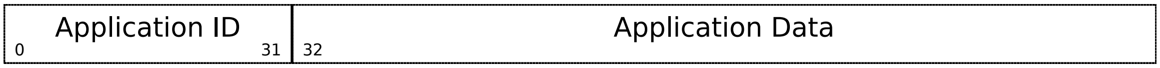
7.2.1 STREAMINFO


Minimum Block Size (in samples)
0
15
Maximum Block Size (in samples)
16
31
Minimum Frame Size (in bytes)
32
55
Maximum Frame Size (in bytes)
56
79
Sample Rate
80
99
Channels
100
102
Bits per Sample
103
107
Total Samples
108
143
MD5SUM of PCM Data
144
271

7.2.2 PADDING

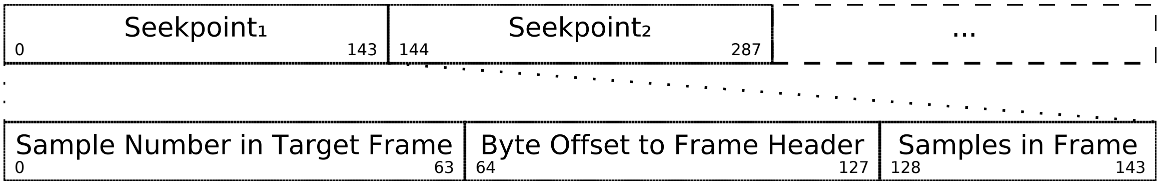
PADDING is simply a block full of NULL (`0x00`) bytes. Its purpose is to provide extra metadata space within the FLAC file. By having a padding block, other metadata blocks can be grown or shrunk without having to rewrite the entire FLAC file by removing or adding space to the padding.

7.2.3 APPLICATION

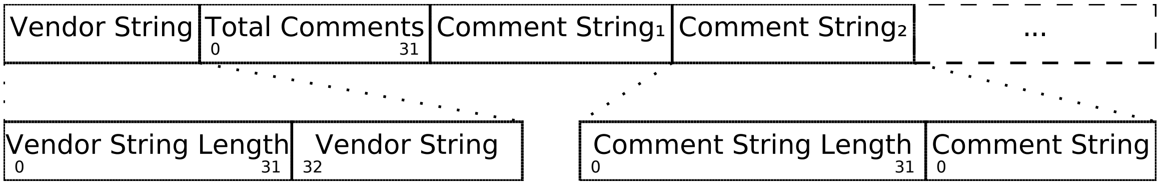


APPLICATION is a general-purpose metadata block used by a variety of different programs. Its contents are defined by the ASCII Application ID value.

7.2.4 SEEKTABLE



7.2.5 VORBIS_COMMENT



The length fields are all little-endian. The Vendor String and Comment Strings are all UTF-8 encoded. Keys are not case-sensitive and may occur multiple times, indicating multiple values for the same field. For instance, a track with multiple artists may have more than one `ARTIST`.

ALBUM album name

ARTIST artist name, band name, composer, author, etc.

CATALOGNUMBER* CD spine number

COMPOSER* the work's author

CONDUCTOR* performing ensemble's leader

COPYRIGHT copyright attribution

DATE recording date

DESCRIPTION a short description

DISCNUMBER* disc number for multi-volume work

ENGINEER* the recording masterer

ENSEMBLE* performing group

GENRE a short music genre label

GUEST ARTIST* collaborating artist

ISRC ISRC number for the track

LICENSE license information

LOCATION recording location

OPUS* number of the work

ORGANIZATION record label

PART* track's movement title

PERFORMER performer name, orchestra, actor, etc.

PRODUCER* person responsible for the project

PRODUCTNUMBER* UPC, EAN, or JAN code

PUBLISHER* album's publisher

RELEASE DATE* date the album was published

REMIXER* person who created the remix

SOURCE ARTIST* artist of the work being performed

SOURCE MEDIUM* CD, radio, cassette, vinyl LP, etc.

SOURCE WORK* a soundtrack's original work

SPARS* DDD, ADD, AAD, etc.

SUBTITLE* for multiple track names in a single file

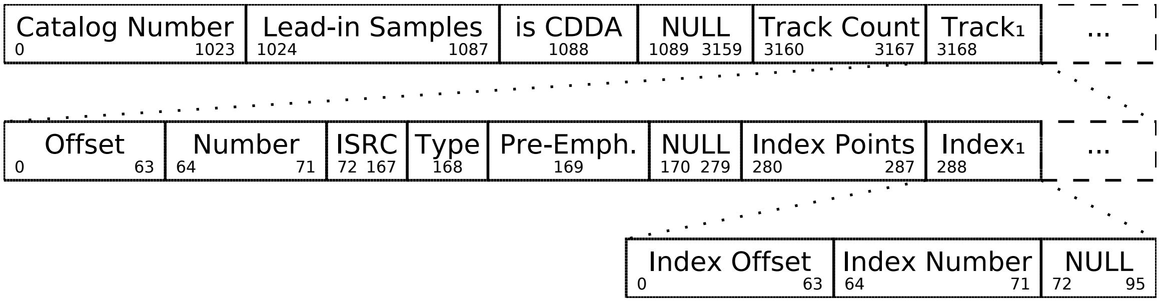
TITLE track name

TRACKNUMBER track number

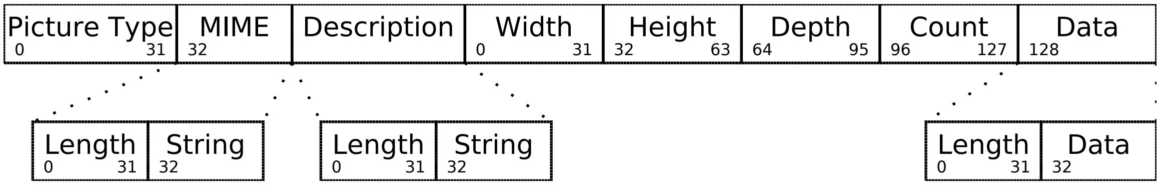
VERSION track version

Fields marked with * are proposed extension fields and not part of the official Vorbis comment specification.

7.2.6 CUESHEET



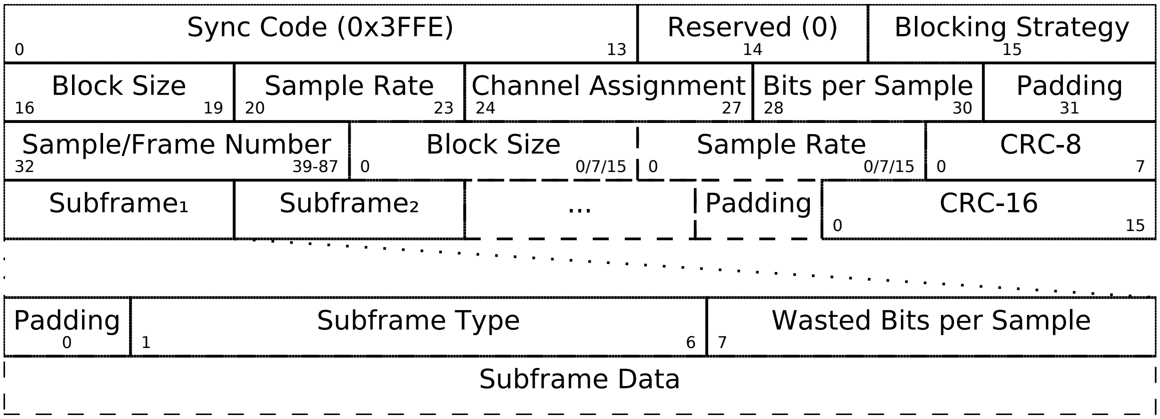
7.2.7 PICTURE



Picture Type	Type
0	Other
1	32x32 pixels ‘file icon’ (PNG only)
2	Other file icon
3	Cover (front)
4	Cover (back)
5	Leaflet page
6	Media (e.g. label side of CD)
7	Lead artist / Lead performer / Soloist
8	Artist / Performer
9	Conductor
10	Band / Orchestra
11	Composer
12	Lyricist / Text writer
13	Recording location
14	During recording
15	During performance
16	Movie / Video screen capture
17	A bright colored fish
18	Illustration
19	Band / Artist logotype
20	Publisher / Studio logotype

7.3 FLAC Decoding

A FLAC stream is made up of individual FLAC frames, as follows:



Value	Block Size	Sample Rate	Channels	Assignment	Bits per Sample	Value
0000	STREAMINFO	STREAMINFO	1	front center	STREAMINFO	0000
0001	192	88200	2	front left, front right	8	0001
0010	576	176400	3	fL, fR, fC	12	0010
0011	1152	192000	4	fL, fR, back left, back right	reserved	0011
0100	2304	8000	5	fL, fR, fC, bL, bR	16	0100
0101	4608	16000	6	fL, fR, fC, LFE, bL, bR	20	0101
0110	8 bits (+1)	22050	7	undefined	24	0110
0111	16 bits (+1)	24000	8	undefined	reserved	0111
1000	256	32000	2	0 left, 1 difference		1000
1001	512	44100	2	0 difference, 1 right		1001
1010	1024	48000	2	0 average, 1 difference		1010
1011	2048	96000		reserved		1011
1100	4096	8 bits (in kHz)		reserved		1100
1101	8192	16 bits (in Hz)		reserved		1101
1110	16384	16 bits (in 10s of Hz)		reserved		1110
1111	32768	invalid		reserved		1111

‘Sample/Frame Number’ is a UTF-8 coded value. If the ‘Blocking Strategy’ is 0, it decodes to a 32-bit frame number. If the ‘Blocking Strategy’ is 1, it decodes to a 36-bit sample number.

There is one ‘Subframe’ per channel.

‘Wasted Bits Per Sample’ is typically a single bit set to 0, indicating no wasted bits per sample. If set to 1, a unary-encoded value follows which indicates how many bits are wasted per sample.

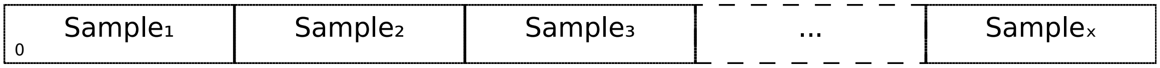
Padding is added as needed between the final subframe and CRC-16 in order to byte-align frames.

Value	Subframe Type
000000	SUBFRAME_CONSTANT
000001	SUBFRAME_VERBATIM
00001x	reserved
0001xx	reserved
001xxx	SUBFRAME_FIXED xxx = predictor order
01xxxx	reserved
1xxxxx	SUBFRAME_LPC xxxxx = predictor order - 1

7.3.1 CONSTANT Subframe

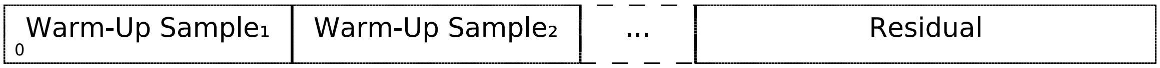
This is the simplest possible subframe. It consists of a single value whose size is equal to the subframe's 'Bits per Sample'[1]. For instance, a 16-bit subframe would have CONSTANT subframes 16 bits in length. The value of the subframe is the value of all samples the subframe contains. An obvious use of this subframe is to store an entire subframe's worth of digital silence (samples with a value of 0) very efficiently.

7.3.2 VERBATIM Subframe



Each sample is the size of the subframe's 'Bits per Sample'. The total number of samples equals the subframe's 'Block Size'. Since it does no compression whatsoever and simply stores audio samples as-is, this subframe is only suitable for especially noisy portions of a track where no suitable predictor can be found.

7.3.3 FIXED Subframe



The number of warm-up samples equals the 'Predictor Order' (which is encoded in the 'Subframe Type'). Each warm-up sample is the same size as the subframe's 'Bits per Sample'. These samples are sent out as-is; they are the subframe's 'starting point' upon which further samples build when decompressing the stream. Determining the value of the current sample is then a matter of looking backwards at previously decoded samples (or warm-up samples), applying a simple formula on their values (which depends on the Predictor Order) and adding the residual.

[1] The subframe's 'Bits per Sample' may not be the same size as the frame's 'Bits per Sample'; if the subframe is a difference or side channel, the subframe's 'Bits per Sample' is 1 bit larger than the frame's. Or, if the subframe has wasted-bits, it is treated as having that many bits fewer. This is explained in more detail on page 49.

For Order 0:

$\text{Sample}_i = \text{Residual}_i$ for $i = 0$ to Block Size $- 1$

For Order 1:

$$\text{Sample}_i = \begin{cases} \text{Warm Up}_i & \text{for } i = 0 \\ \text{Sample}_{i-1} + \text{Residual}_{i-1} & \text{for } i = 1 \text{ to Block Size} - 1 \end{cases}$$

For Order 2:

$$\text{Sample}_i = \begin{cases} \text{Warm Up}_i & \text{for } i = 0 \text{ to } 1 \\ (2 \times \text{Sample}_{i-1}) - \text{Sample}_{i-2} + \text{Residual}_{i-2} & \text{for } i = 2 \text{ to Block Size} - 1 \end{cases}$$

For Order 3:

$$\text{Sample}_i = \begin{cases} \text{Warm Up}_i \\ \ \text{for } i = 0 \text{ to } 2 \\ (3 \times \text{Sample}_{i-1}) - (3 \times \text{Sample}_{i-2}) + \text{Sample}_{i-3} + \text{Residual}_{i-3} \\ \ \text{for } i = 3 \text{ to Block Size} - 1 \end{cases}$$

For Order 4:

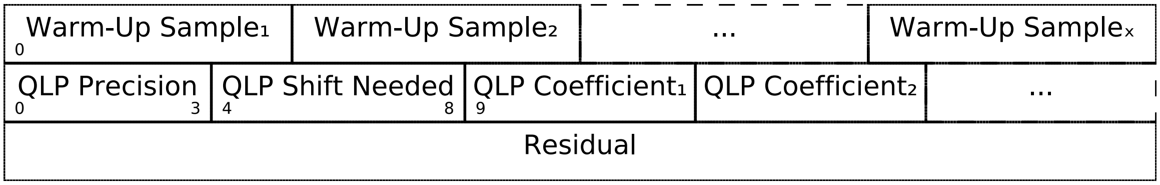
$$\text{Sample}_i = \begin{cases} \text{Warm Up}_i \\ \ \text{for } i = 0 \text{ to } 3 \\ (4 \times \text{Sample}_{i-1}) - (6 \times \text{Sample}_{i-2}) + (4 \times \text{Sample}_{i-3}) - \text{Sample}_{i-4} + \text{Residual}_{i-4} \\ \ \text{for } i = 4 \text{ to Block Size} - 1 \end{cases}$$

For example, given a FIXED order of 1, a Warm Up_0 of 10, the residuals 1, 2, -2, 1 -1, and a 'Block Size' of 6, we calculate samples as follows:

$$\begin{aligned} \text{Sample}_0 &= \text{Warm Up}_0 = \mathbf{10} \\ \text{Sample}_1 &= \text{Sample}_0 + \text{Residual}_0 = 10 + 1 = \mathbf{11} \\ \text{Sample}_2 &= \text{Sample}_1 + \text{Residual}_1 = 11 + 2 = \mathbf{13} \\ \text{Sample}_3 &= \text{Sample}_2 + \text{Residual}_2 = 13 - 2 = \mathbf{11} \\ \text{Sample}_4 &= \text{Sample}_3 + \text{Residual}_3 = 11 + 1 = \mathbf{12} \\ \text{Sample}_5 &= \text{Sample}_4 + \text{Residual}_4 = 12 - 1 = \mathbf{11} \end{aligned}$$

How to extract the encoded residual is explained on page 46.

7.3.4 LPC Subframe



The number of warm-up samples equals the 'LPC Order' (which is encoded in the 'Subframe Type'). The size if each warm-up sample equals the subframe's 'Bits per Sample'. The size of each QLP Coefficient is equal to 'QLP Precision' number of bits, plus 1. 'QLP Shift Needed' and the value of each Coefficient are signed two's-complement integers. The number of Coefficients equals the 'LPC Order'.

For $i = 0$ to $Order - 1$:

$$\text{Sample}_i = \text{Warm Up}_i$$

For $i = Order$ to Block Size - 1:

$$\text{Sample}_i = \left\lfloor \frac{\sum_{j=0}^{Order-1} \text{LPC Coefficient}_j \times \text{Sample}_{i-j-1}}{2^{\text{QLP Shift Needed}}} \right\rfloor + \text{Residual}_{i-Order}$$

This simply means we're taking the sum of the calculated values from 0 to Order - 1, bit-shifting that sum down and added the residual when determining the current sample. Much like the FIXED subframe, LPC subframes also contain warm-up samples which serve as our calculation's starting point.

In this example, the 'LPC Order' is 5, the 'QLP Shift Needed' is 9, the 'Block Size' is 10 and the remaining subframe values are as follows:

Warm Up_0	1053	LPC Coefficient_0	1241	Residual_0	11
Warm Up_1	1116	LPC Coefficient_1	-944	Residual_1	79
Warm Up_2	1257	LPC Coefficient_2	14	Residual_2	24
Warm Up_3	1423	LPC Coefficient_3	342	Residual_3	-81
Warm Up_4	1529	LPC Coefficient_4	-147	Residual_4	-72

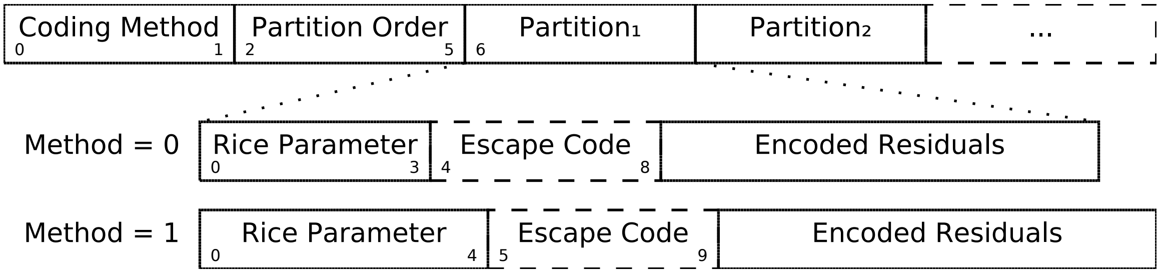
Note that the number is residuals equals Block Size minus Order which happens to be 5 in this simple example ($10 - 5 = 5$). Most of the time it will be much larger.

For the sake of brevity, in this example we'll abbreviate 'QLP Coefficient' as 'C' and 'Sample' as 'S'.

$$
\begin{aligned}
\text{Sample}_0 &= \text{Warm Up}_0 = \mathbf{1053} \\
\text{Sample}_1 &= \text{Warm Up}_1 = \mathbf{1116} \\
\text{Sample}_2 &= \text{Warm Up}_2 = \mathbf{1257} \\
\text{Sample}_3 &= \text{Warm Up}_3 = \mathbf{1423} \\
\text{Sample}_4 &= \text{Warm Up}_4 = \mathbf{1529} \\
\text{Sample}_5 &= \left\lfloor \frac{(C_0 \times S_4) + (C_1 \times S_3) + (C_2 \times S_2) + (C_3 \times S_1) + (C_4 \times S_0)}{2^9} \right\rfloor + \text{Residual}_0 \\
&= \left\lfloor \frac{(1241 \times 1529) + (-944 \times 1423) + (14 \times 1257) + (342 \times 1116) + (-147 \times 1053)}{512} \right\rfloor + 11 \\
&= \left\lfloor \frac{798656}{512} \right\rfloor + 11 = 1559 + 11 = \mathbf{1570} \\
\text{Sample}_6 &= \left\lfloor \frac{(C_0 \times S_5) + (C_1 \times S_4) + (C_2 \times S_3) + (C_3 \times S_2) + (C_4 \times S_1)}{2^9} \right\rfloor + \text{Residual}_1 \\
&= \left\lfloor \frac{(1241 \times 1570) + (-944 \times 1529) + (14 \times 1423) + (342 \times 1257) + (-147 \times 1116)}{512} \right\rfloor + 79 \\
&= \left\lfloor \frac{790758}{512} \right\rfloor + 79 = 1544 + 79 = \mathbf{1623} \\
\text{Sample}_7 &= \left\lfloor \frac{(C_0 \times S_6) + (C_1 \times S_5) + (C_2 \times S_4) + (C_3 \times S_3) + (C_4 \times S_2)}{2^9} \right\rfloor + \text{Residual}_2 \\
&= \left\lfloor \frac{(1241 \times 1623) + (-944 \times 1570) + (14 \times 1529) + (342 \times 1423) + (-147 \times 1257)}{512} \right\rfloor + 24 \\
&= \left\lfloor \frac{855356}{512} \right\rfloor + 24 = 1670 + 24 = \mathbf{1694} \\
\text{Sample}_8 &= \left\lfloor \frac{(C_0 \times S_7) + (C_1 \times S_6) + (C_2 \times S_5) + (C_3 \times S_4) + (C_4 \times S_3)}{2^9} \right\rfloor + \text{Residual}_3 \\
&= \left\lfloor \frac{(1241 \times 1694) + (-944 \times 1623) + (14 \times 1570) + (342 \times 1529) + (-147 \times 1423)}{512} \right\rfloor - 81 \\
&= \left\lfloor \frac{1769}{512} \right\rfloor - 81 = 1769 - 81 = \mathbf{1688} \\
\text{Sample}_9 &= \left\lfloor \frac{(C_0 \times S_8) + (C_1 \times S_7) + (C_2 \times S_6) + (C_3 \times S_5) + (C_4 \times S_4)}{2^9} \right\rfloor + \text{Residual}_4 \\
&= \left\lfloor \frac{(1241 \times 1688) + (-944 \times 1694) + (14 \times 1623) + (342 \times 1570) + (-147 \times 1529)}{512} \right\rfloor - 72 \\
&= \left\lfloor \frac{830571}{512} \right\rfloor - 72 = 1622 - 72 = \mathbf{1550}
\end{aligned}
$$

7.3.5 the Residual

Though the FLAC format allows for different forms of residual coding, two forms of partitioned Rice are the only ones currently supported. The difference between the two is that when ‘Coding Method’ is 0, the Rice Parameter in each partition is 4 bits. When the ‘Coding Method’ is 1, that parameter is 5 bits.



There are $2^{\text{Partition Order}}$ number of Partitions. The number of decoded residuals in a Partition depends on the its position in the subframe. The first partition in the subframe contains:

$$\text{Total Residuals} = \frac{\text{Frame's Block Size}}{2^{\text{Partition Order}}} - \text{Predictor Order}$$

Subsequent partitions contain:

$$\text{Total Residuals} = \frac{\text{Frame's Block Size}}{2^{\text{Partition Order}}}$$

Unless the Partition Order is 0. In that case:

$$\text{Total Residuals} = \text{Frame's Block Size} - \text{Predictor Order}$$

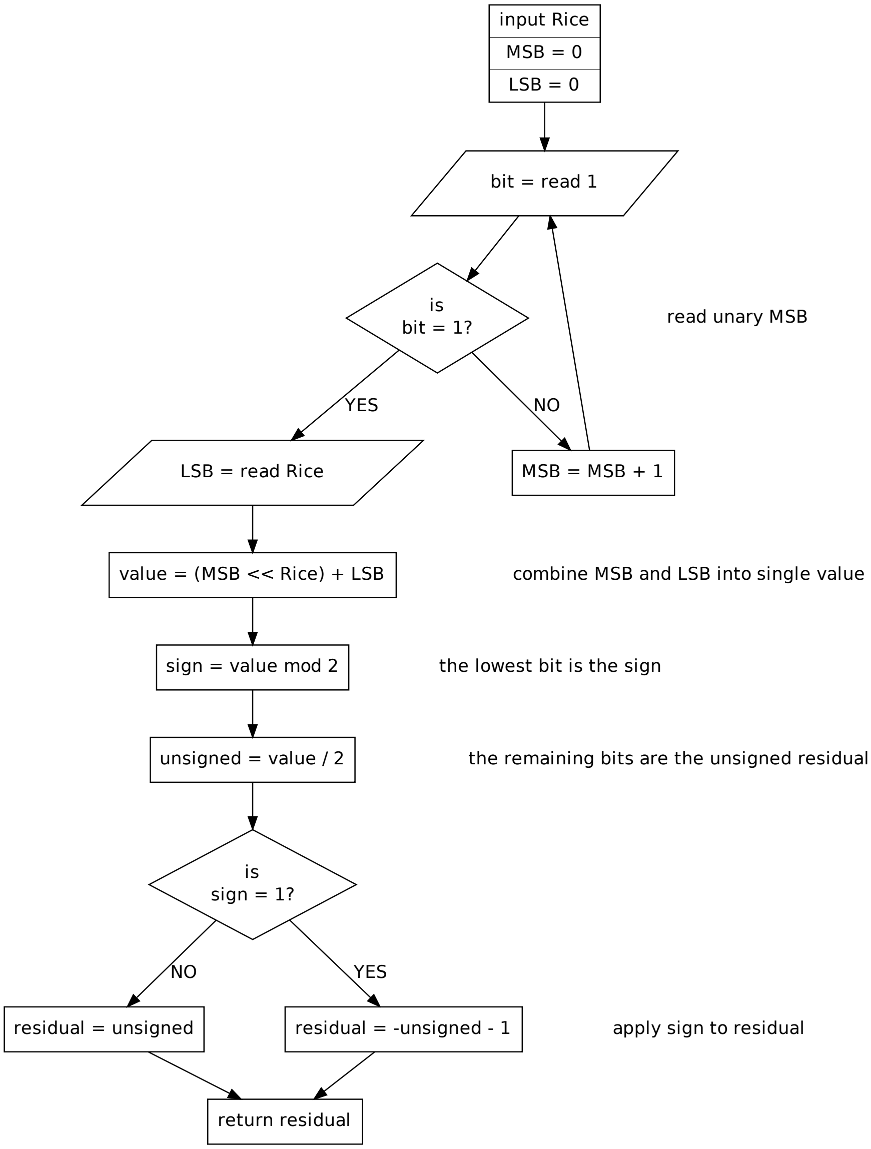
since there is only one partition which takes up the entire block.

Why does the subframe’s block of residuals always have ‘Predictor Order’ less residuals than the frame’s block size? The reason is that FIXED or LPC subframes have ‘Predictor Order’ number of unencoded warm-up samples. So, the total number of warm-up samples and the total number residuals adds up to the frame’s block size.

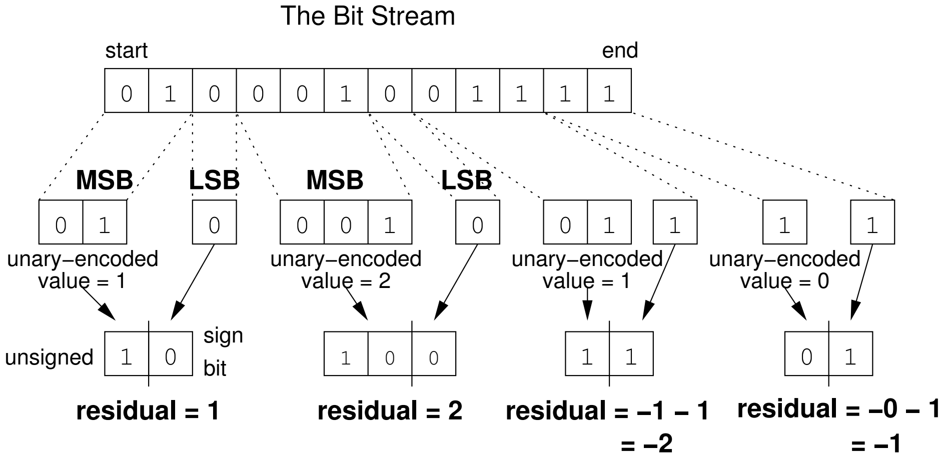
If all of the bits in ‘Rice Parameter’ are set, the partition is unencoded binary using ‘Escape Code’ number of bits per sample.

Rice Encoding

The residual uses Rice coding to compress lots of mostly small values in a very small amount of space. Decoding the residuals requires the Rice parameter and may be easier to explain with a flowchart of the process, which is repeated for each residual in the partition:

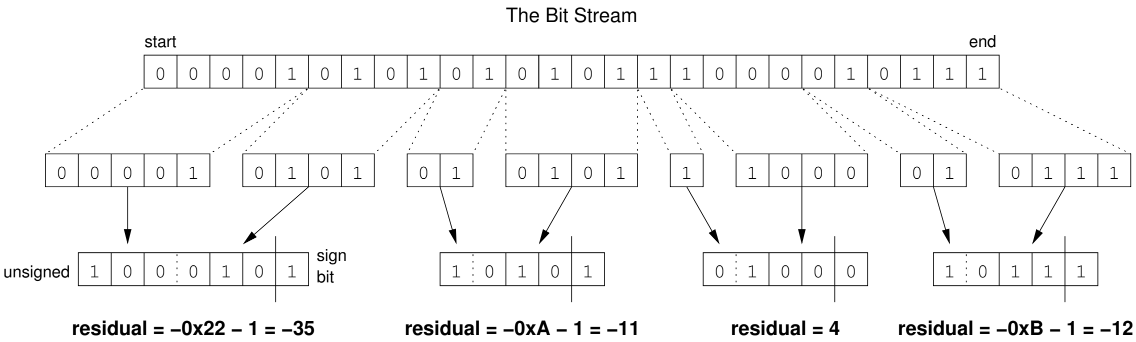


Lets run through an example in which the Rice parameter is 1, indicating 1 least-significant bit per residual value:



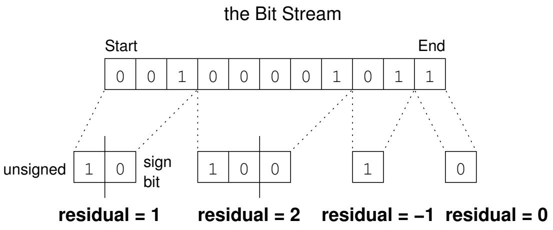
Thus, our final decoded residual values are 1, 2, -2 and -1 and have consumed a total of 12 bits from the stream.

Next, lets run through an example in which the Rice parameter is 4:



In this case, our final residual values are -35, -11, 4 and -12.

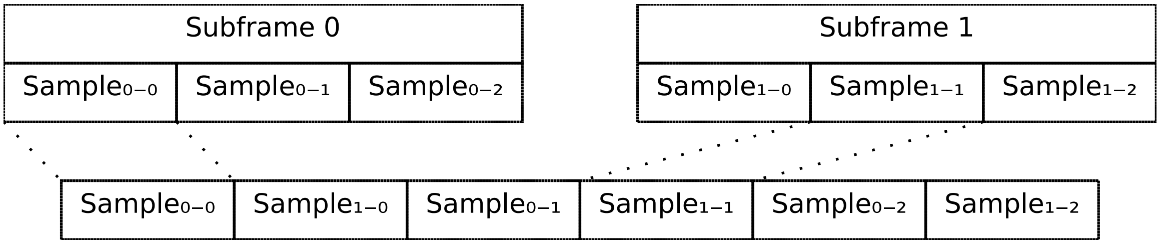
Finally, we'll try a case in which the Rice parameter is 0, indicating no least-significant bits at all:



Our final residual values are 1, 2, -1 and 0.

7.3.6 Channel Assignment

Since most audio has more than one channel, it is important to understand how FLAC handles putting it back together. When channels are stored independently, one simply interleaves them together in the proper order. Let's take an example of 2 channel, 16-bit audio stored this way:



This is the simplest case. However, in the case of difference or mid-side channels, one subframe will contain actual channel data and the other channel will contain signed difference data which is applied to that actual data in order to reconstruct both channels. It's very important to remember that the difference (and side) channel has 1 additional bit per sample which will be consumed during reconstruction. Why 1 additional bit? Let's take an example where the left sample's value is -30000 and the right sample's value is +30000. Storing this pair as left + difference means the left sample remains -30000 and the difference is -60000 ($-30000 - -60000 = +30000$). -60000 won't fit into a 16-bit signed integer. Adding that 1 additional bit doubles our range of values and that's just enough to cover any possible difference between two samples.

Performing the actual channel decorrelation is as follows:

Left-Difference (assignment `0x8`):

$$\begin{aligned} \text{Left}_i &= \text{Subframe}_{0\ i} \\ \text{Right}_i &= \text{Subframe}_{0\ i} - \text{Subframe}_{1\ i} \end{aligned}$$

Difference-Right (assignment `0x9`):

$$\begin{aligned} \text{Left}_i &= \text{Subframe}_{0\ i} + \text{Subframe}_{1\ i} \\ \text{Right}_i &= \text{Subframe}_{1\ i} \end{aligned}$$

Mid-Side (assignment `0xA`):

$$\begin{aligned} \text{Left}_i &= \lfloor(((\text{Subframe}_{0\ i} \times 2) + (\text{Subframe}_{1\ i} \bmod 2)) + \text{Subframe}_{1\ i}) \div 2\rfloor \\ \text{Right}_i &= \lfloor(((\text{Subframe}_{0\ i} \times 2) + (\text{Subframe}_{1\ i} \bmod 2)) - \text{Subframe}_{1\ i}) \div 2\rfloor \end{aligned}$$

Let's take an example of two Mid-Side encoded samples: 1533 and 3039 for Subframe$_0$ and Subframe$_1$, respectively:

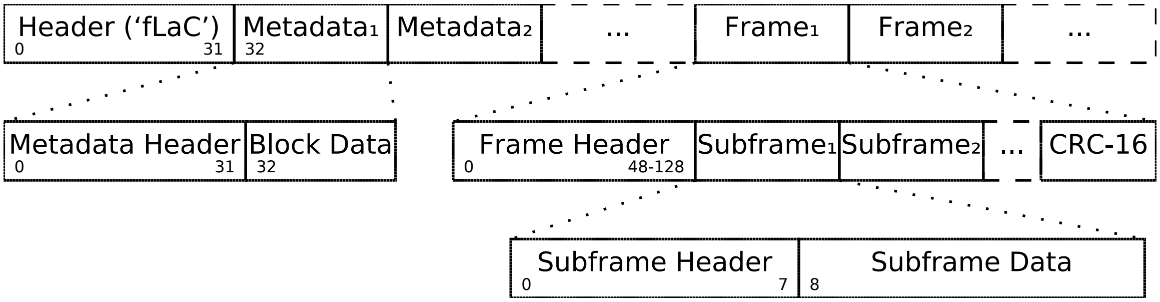
$$\begin{aligned}
\text{Left} &= \lfloor(((1533 \times 2) + (3039 \bmod 2)) + 3039) \div 2\rfloor \\
&= \lfloor((3066 + 1) + 3039) \div 2\rfloor \\
&= \lfloor 6106 \div 2\rfloor = \mathbf{3053} \\
\text{Right} &= \lfloor(((1533 \times 2) + (3039 \bmod 2)) - 3039) \div 2\rfloor \\
&= \lfloor((3066 + 1) - 3039) \div 2\rfloor \\
&= \lfloor 28 \div 2\rfloor = \mathbf{14}
\end{aligned}$$

7.3.7 Wasted Bits per Sample

Though rare in practice, FLAC subframes support 'wasted bits per sample'. Put simply, these wasted bits are removed during subframe calculation and restored to the subframe's least significant bits as zero value bits when it is returned. For instance, a subframe with 1 wasted bit per sample in a 16-bit FLAC stream is treated as having only 15 bits per sample when reading warm-up samples and then all through the rest of the subframe calculation. That wasted zero bit is then prepended to each sample prior to returning the subframe.

7.4 FLAC Encoding

For the purposes of discussing FLAC encoding, we'll assume one has a stream of input PCM values along with the stream's sample rate, number of channels and bits per sample. Creating a valid FLAC file is then a matter of writing the proper file header, metadata blocks and FLAC frames.



bits	value
1	0 if additional metadata blocks follow, 1 if not
7	0 for STREAMINFO, 1 for PADDING, 4 for VORBIS_COMMENT, etc.
24	the length of the block data in bytes, not including the header

Figure 7.1: Metadata Header

7.4.1 the STREAMINFO Metadata Block

bits	value
16	the minimum FLAC frame size, in PCM frames
16	the maximum FLAC frame size, in PCM frames
24	the minimum FLAC frame size, in bytes
24	the maximum FLAC frame size, in bytes
20	the stream's sample rate, in Hz
3	the stream's channel count, minus one
5	the stream's bit-per-sample, minus one
36	the stream's total number of PCM frames
128	an MD5 sum of the PCM stream's bytes

This metadata block must come first and is the only required block in a FLAC file.

When encoding a FLAC file, many of these fields cannot be known in advance. Instead, one must keep track of those values during encoding and then rewrite the STREAMINFO block when finished.

7.4.2 the Frame Header

bits	value
14	`0x3FFE` sync code
1	`0` reserved
1	`0` if the header encodes the frame number, `1` if it encodes the sample number
4	this frame's block size, as encoded PCM frames
4	this frame's encoded sample rate
4	this frame's encoded channel assignment
3	this frame's encoded bits per sample
1	`0` padding
8-56	the frame number, or sample number, UTF-8 encoded and starting from 0
0/8/16	the number of PCM frames (minus one) in this FLAC frame if block size is `0x6` (8 bits) or `0x7` (16 bits)
0/8/16	the sample rate of this FLAC frame if sample rate is `0xC` (8 bits), `0xD` (16 bits) or `0xE` (16 bits)
8	the CRC-8 of all data from the beginning of the frame header

The FLAC frame's block size in PCM frames (called "channel independent samples" in FLAC's documentation) is typically encoded in the 4 bit 'block size' field. But for odd-sized frames - which often occur at the end of the stream - that value is stored as an 8 or 16 bit integer following the UTF-8 encoded frame number.

In addition, odd sample rate values are stored as 8 bit (in kHz), 16 bit (in Hz) or 16 bit (in 10s of Hz) prior to the CRC-8, should a predefined value not be available.

Up until this point, nearly all of these fields can be filled from the PCM stream data. Unless you're writing a variable block size encoder, one should encode the frame number starting from 0 in the frame header and choose a predefined block size for as many FLAC frames as possible.

7.4.3 Channel Assignment

If the input stream has a number of channels other than 2, one has no choice but to store them independently. If the number of channels equals 2, one can try all four possible assignments and use the one which takes the least amount of space.

Left-Difference (assignment `0x8`):

$$\text{Subframe}_{0\ i} = \text{Left}_i$$
$$\text{Subframe}_{1\ i} = \text{Left}_i - \text{Right}_i$$

Difference-Right (assignment `0x9`):

$$\text{Subframe}_{0\ i} = \text{Left}_i - \text{Right}_i$$
$$\text{Subframe}_{1\ i} = \text{Right}_i$$

Mid-Side (assignment `0xA`):

$$\text{Subframe}_{0\ i} = \lfloor(\text{Left}_i + \text{Right}_i) \div 2\rfloor$$
$$\text{Subframe}_{1\ i} = \text{Left}_i - \text{Right}_i$$

Remember that the difference and side channels are treated as having 1 additional bit during encoding. Thus, a 16 bit frame would have a 17 bit difference subframe when calculating warm-up samples.

7.4.4 the Subframe Header

bits	value
1	`0` padding
`000000`	`SUBFRAME_CONSTANT`
`000001`	`SUBFRAME_VERBATIM`
`001xxx`	`SUBFRAME_FIXED` (`xxx` = Predictor Order)
`1xxxxx`	`SUBFRAME_LPC` (`xxxxx` = Predictor Order - 1)
1	`0` if no wasted bits per sample, `1` if a unary-encoded number follows
0+	the number of wasted bits per sample (minus one) encoded as unary

7.4.5 the CONSTANT Subframe

If all the samples in a subframe are identical, one can encode them using a `CONSTANT` subframe.

7.4.6 the VERBATIM Subframe

This subframe simply stores all the samples as-is, with no compression whatsoever.

7.4.7 the FIXED Subframe

This subframe consists of ‘predictor order’ number of unencoded warm-up samples followed by a residual. Determining which predictor order to use on a given set of input samples depends on their minimum delta sum. This process is best explained by example:

index	sample	Δ^0	Δ^1	Δ^2	Δ^3	Δ^4
0	-40					
1	-41	*-41*				
2	-40	*-40*	*-1*			
3	-39	*-39*	*-1*	*0*		
4	-38	-38	*-1*	*0*	*0*	
5	-38	-38	0	-1	1	-1
6	-35	-35	-3	3	-4	5
7	-35	-35	0	-3	6	-10
8	-39	-39	4	-4	1	5
9	-40	-40	1	3	-7	8
10	-40	-40	0	1	2	-9
11	-39	-39	-1	1	0	2
12	-38	-38	-1	0	1	-1
13	-37	-37	-1	0	0	1
14	-33	-33	-4	3	-3	3
15	-36	-36	3	-7	10	-13
16	-35	-35	-1	4	-11	21
17	-31	-31	-4	3	1	-12
18	-32	-32	1	-5	8	-7
19	-33	-33	1	0	-5	13
$\lvert sum \rvert$		579	26	38	60	111

Note that the numbers in italics play a part in the delta calculation to their right, but do **not** figure into the delta’s absolute value sum, below.

In this example, Δ^1’s value of 26 is the smallest. Therefore, when compressing this set of samples in a FIXED subframe, it’s best to use a predictor order of 1.

The predictor order indicates how many warm-up samples to take from the PCM stream. Determining the residual values can then be done automatically based on the current Sample_i and previously encoded samples, or warm-up samples.

The residual encoding process is then the simple inverse of the decoding process, as follows:

For Order 0:

$$\begin{aligned}\text{Residual}_i &= \text{Sample}_i\\ &\quad\text{for } i = 0 \text{ to Block Size} - 1\end{aligned}$$

For Order 1:

$$\begin{aligned}\text{Warm Up}_0 &= \text{Sample}_0\\ \text{Residual}_i &= \text{Sample}_{i+1} - \text{Sample}_i\\ &\quad\text{for } i = 0 \text{ to Block Size} - 2\end{aligned}$$

For Order 2:

$$\begin{aligned}\text{Warm Up}_j &= \text{Sample}_j\\ &\quad\text{for } j = 0 \text{ to } 1\\ \text{Residual}_i &= \text{Sample}_{i+2} - ((2 \times \text{Sample}_{i+1}) - \text{Sample}_i)\\ &\quad\text{for } i = 0 \text{ to Block Size} - 3\end{aligned}$$

For Order 3:

$$\begin{aligned}\text{Warm Up}_j &= \text{Sample}_j\\ &\quad\text{for } j = 0 \text{ to } 2\\ \text{Residual}_i &= \text{Sample}_{i+3} - ((3 \times \text{Sample}_{i+2}) - (3 \times \text{Sample}_{i+1}) + \text{Sample}_i)\\ &\quad\text{for } i = 0 \text{ to Block Size} - 4\end{aligned}$$

For Order 4:

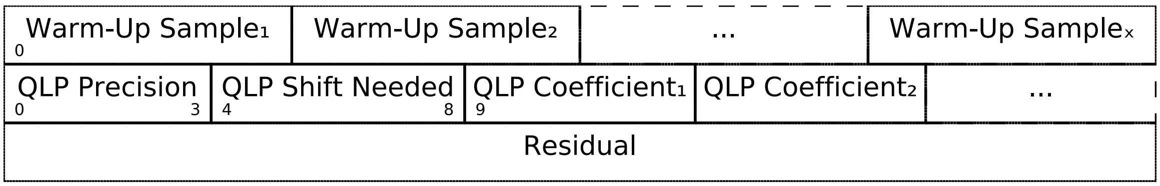
$$\begin{aligned}\text{Warm Up}_j &= \text{Sample}_j\\ &\quad\text{for } j = 0 \text{ to } 3\\ \text{Residual}_i &= \text{Sample}_{i+4} - ((4 \times \text{Sample}_{i+3}) - (6 \times \text{Sample}_{i+2}) + (4 \times \text{Sample}_{i+1}) - \text{Sample}_i)\\ &\quad\text{for } i = 0 \text{ to Block Size} - 5\end{aligned}$$

So to complete our FIXED subframe encoding example in which the predictor order is 1:

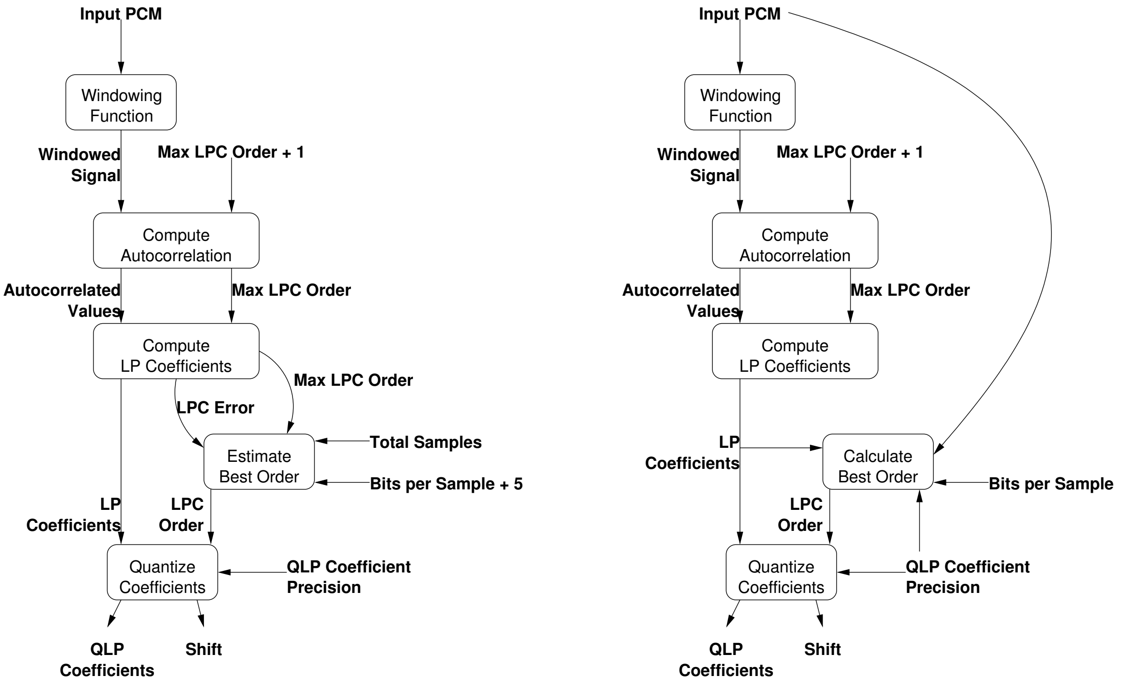
$$\begin{aligned}
\text{Warm Up}_0 &= \text{Sample}_0 = \textbf{-40}\\
\text{Residual}_0 &= \text{Sample}_1 - \text{Sample}_0 = -41 - -40 = \textbf{-1}\\
\text{Residual}_1 &= \text{Sample}_2 - \text{Sample}_1 = -40 - -41 = \textbf{1}\\
\text{Residual}_2 &= \text{Sample}_3 - \text{Sample}_2 = -39 - -40 = \textbf{1}\\
\text{Residual}_3 &= \text{Sample}_4 - \text{Sample}_3 = -38 - -39 = \textbf{1}\\
\text{Residual}_4 &= \text{Sample}_5 - \text{Sample}_4 = -38 - -38 = \textbf{0}\\
\text{Residual}_5 &= \text{Sample}_6 - \text{Sample}_5 = -35 - -38 = \textbf{3}\\
\text{Residual}_6 &= \text{Sample}_7 - \text{Sample}_6 = -35 - -35 = \textbf{0}\\
\text{Residual}_7 &= \text{Sample}_8 - \text{Sample}_7 = -39 - -35 = \textbf{-4}\\
\text{Residual}_8 &= \text{Sample}_9 - \text{Sample}_8 = -40 - -39 = \textbf{-1}\\
\text{Residual}_9 &= \text{Sample}_{10} - \text{Sample}_9 = -40 - -40 = \textbf{0}\\
\text{Residual}_{10} &= \text{Sample}_{11} - \text{Sample}_{10} = -39 - -40 = \textbf{1}\\
\text{Residual}_{11} &= \text{Sample}_{12} - \text{Sample}_{11} = -38 - -39 = \textbf{1}\\
\text{Residual}_{12} &= \text{Sample}_{13} - \text{Sample}_{12} = -37 - -38 = \textbf{1}\\
\text{Residual}_{13} &= \text{Sample}_{14} - \text{Sample}_{13} = -33 - -37 = \textbf{4}\\
\text{Residual}_{14} &= \text{Sample}_{15} - \text{Sample}_{14} = -36 - -33 = \textbf{-3}\\
\text{Residual}_{15} &= \text{Sample}_{16} - \text{Sample}_{15} = -35 - -36 = \textbf{1}\\
\text{Residual}_{16} &= \text{Sample}_{17} - \text{Sample}_{16} = -31 - -35 = \textbf{4}\\
\text{Residual}_{17} &= \text{Sample}_{18} - \text{Sample}_{17} = -32 - -31 = \textbf{-1}\\
\text{Residual}_{18} &= \text{Sample}_{19} - \text{Sample}_{18} = -33 - -32 = \textbf{-1}
\end{aligned}$$

7.4.8 the LPC Subframe

Unlike the FIXED subframe which required only input samples and a predictor order, LPC subframes also require a list of QLP coefficients, a QLP precision value of those coefficients, and a QLP shift needed value.



Determining these values for a given input PCM signal is a somewhat complicated process which depends on whether one is performing an exhaustive LP coefficient order search or not:



(a) non-exhaustive search

(b) exhaustive search

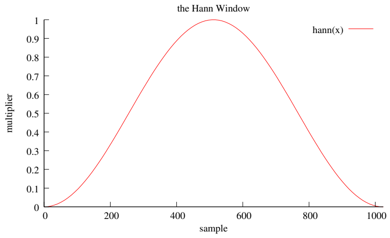
Windowing

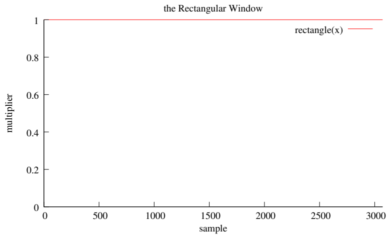
The first step in LPC subframe encoding is ‘windowing’ the input signal. Put simply, this is a process of multiplying each input sample by an equivalent value from the window, which are floats from 0.0 to 1.0. In this case, the default is a Tukey window with a ratio of 0.5. A Tukey window is a combination of the Hann and Rectangular windows. The ratio of 0.5 means there’s 0.5 samples in the Hann window per sample in the Rectangular window.

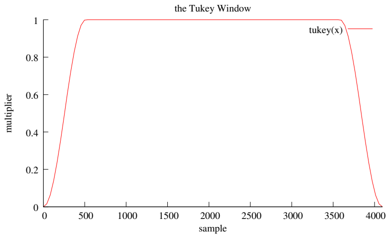
$$\text{hann}(n) = \frac{1}{2}\left(1 - \cos\left(\frac{2\pi n}{\text{sample count} - 1}\right)\right) \tag{7.1}$$

$$\text{rectangle}(n) = 1.0 \tag{7.2}$$

The Tukey window is defined by taking a Hann window, splitting it at the halfway point, and inserting a Rectangular window between the two.





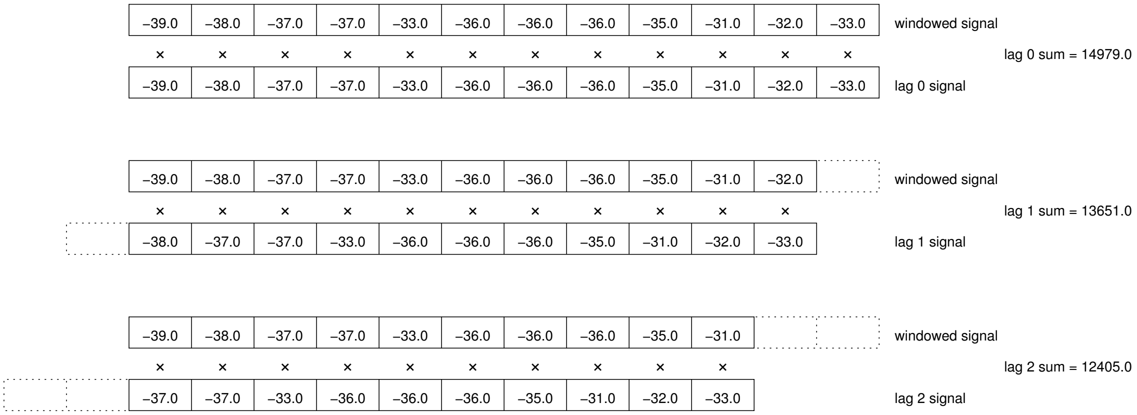


Let’s run through a short example with 20 samples:

index	input sample		Tukey window		windowed signal
0	-40	×	0.0000	=	0.00
1	-41	×	0.1464	=	-6.00
2	-40	×	0.5000	=	-20.00
3	-39	×	0.8536	=	-33.29
4	-38	×	1.0000	=	-38.00
5	-38	×	1.0000	=	-38.00
6	-35	×	1.0000	=	-35.00
7	-35	×	1.0000	=	-35.00
8	-39	×	1.0000	=	-39.00
9	-40	×	1.0000	=	-40.00
10	-40	×	1.0000	=	-40.00
11	-39	×	1.0000	=	-39.00
12	-38	×	1.0000	=	-38.00
13	-37	×	1.0000	=	-37.00
14	-33	×	1.0000	=	-33.00
15	-36	×	1.0000	=	-36.00
16	-35	×	0.8536	=	-29.88
17	-31	×	0.5000	=	-15.50
18	-32	×	0.1464	=	-4.68
19	-33	×	0.0000	=	0.00

Computing Autocorrelation

Once our input samples have been converted to a windowed signal, we then compute the autocorrelation values from that signal. Each autocorrelation value is determined by multiplying the signal's samples by the samples of a lagged version of that same signal, and then taking the sum. The lagged signal is simply the original signal with 'lag' number of samples removed from the beginning.


–39.0 –38.0 –37.0 –37.0 –33.0 –36.0 –36.0 –36.0 –35.0 –31.0 –32.0 –33.0 windowed signal
× × × × × × × × × × × ×
lag 0 sum = 14979.0
–39.0 –38.0 –37.0 –37.0 –33.0 –36.0 –36.0 –36.0 –35.0 –31.0 –32.0 –33.0 lag 0 signal
–39.0 –38.0 –37.0 –37.0 –33.0 –36.0 –36.0 –36.0 –35.0 –31.0 –32.0 windowed signal
× × × × × × × × × × ×
lag 1 sum = 13651.0
–38.0 –37.0 –37.0 –33.0 –36.0 –36.0 –36.0 –35.0 –31.0 –32.0 –33.0 lag 1 signal
–39.0 –38.0 –37.0 –37.0 –33.0 –36.0 –36.0 –36.0 –35.0 –31.0 windowed signal
× × × × × × × × × ×
lag 2 sum = 12405.0
–37.0 –37.0 –33.0 –36.0 –36.0 –36.0 –35.0 –31.0 –32.0 –33.0 lag 2 signal

The lagged sums from 0 to the maximum LPC order are our autocorrelation values. In this example, they are 14979.0, 13651.0 and 12405.0.

LP Coefficient Calculation

Calculating the LP coefficients uses the Levinson-Durbin recursive method.[2] Our inputs are M, the maximum LPC order minus 1, and r autocorrelation values, from $r(0)$ to $r(M-1)$. Our outputs are a, a list of LP coefficient lists from a_{11} to $a_{(M-1)(M-1)}$, and E, a list of error values from E_0 to $E_{(M-1)}$. q_m and κ_m are temporary values.

Initial values:

$$E_0 = r(0) \tag{7.3}$$

$$a_{11} = \kappa_1 = \frac{r(1)}{E_0} \tag{7.4}$$

$$E_1 = E_0(1 - {\kappa_1}^2) \tag{7.5}$$

[2] This algorithm is taken from `http://www.engineer.tamuk.edu/SPark/chap7.pdf`

With $m \geq 2$, the following recursive algorithm is performed:

Step 1. $$q_m = r(m) - \sum_{i=1}^{m-1} a_{i(m-1)} r(m-i) \tag{7.6}$$

Step 2. $$\kappa_m = \frac{q_m}{E_{(m-1)}} \tag{7.7}$$

Step 3. $$a_{mm} = \kappa_m \tag{7.8}$$

Step 4. $$a_{im} = a_{i(m-1)} - \kappa_m a_{(m-i)(m-1)} \text{ for } i = 1,\ i = 2, \ldots, i = m-1 \tag{7.9}$$

Step 5. $$E_m = E_{m-1}(1 - {\kappa_m}^2) \tag{7.10}$$

Step 6. $$\text{If } m < M \text{ then } m \leftarrow m+1 \text{ and goto step 1. If } m = M \text{ then stop.} \tag{7.11}$$

Let's run through an example in which $M = 4$, $r(0) = 11018$, $r(1) = 9690$, $r(2) = 8443$ and $r(3) = 7280$:

$$
\begin{aligned}
E_0 &= r(0) = 11018 \\
a_{11} &= \kappa_1 = \frac{r(1)}{E_0} = \frac{9690}{11018} = 0.8795 \\
E_1 &= E_o(1 - {\kappa_1}^2) = 11018(1 - 0.8795^2) = 2495 \\
q_2 &= r(2) - \sum_{i=1}^{1} a_{i1} r(2-i) = 8443 - (0.8795)(9690) = -79.35 \\
\kappa_2 &= \frac{q_2}{E_1} = \frac{-79.35}{2495} = -0.0318 \\
a_{22} &= \kappa_2 = -0.0318 \\
a_{12} &= a_{11} - \kappa_2 a_{11} = 0.8795 - (-0.0318)(0.8795) = 0.9074 \\
E_2 &= E_1(1 - {\kappa_2}^2) = 2495(1 - -0.0318^2) = 2492 \\
q_3 &= r(3) - \sum_{i=1}^{2} a_{i2} r(3-i) = 7280 - ((0.9074)(8443) + (-0.0318)(9690)) = -73.04 \\
\kappa_3 &= \frac{q_3}{E_2} = \frac{-73.04}{2492} = -0.0293 \\
a_{33} &= \kappa_3 = -0.0293 \\
a_{13} &= a_{12} - \kappa_3 a_{22} = 0.9074 - (-0.0293)(-0.0318) = 0.9065 \\
a_{23} &= a_{22} - \kappa_3 a_{12} = -0.0318 - (-0.0293)(0.9074) = -0.0052 \\
E_3 &= E_2(1 - {\kappa_3}^2) = 2492(1 - -0.0293^2) = 2490
\end{aligned}
$$

Our final values are:

$$
\begin{array}{lll}
a_{11} = 0.8795 & & \\
a_{12} = 0.9074 & a_{22} = -0.0318 & \\
a_{13} = 0.9065 & a_{23} = -0.0052 & a_{33} = -0.0293 \\
E_1 = 2495 & E_2 = 2492 & E_3 = 2490
\end{array}
$$

These values have been rounded to the nearest significant digit and will not be an exact match to those generated by a computer.

Best Order Estimation

At this point, we have an array of prospective LP coefficient lists, a list of error values and must decide which LPC order to use. There are two ways to accomplish this: we can either estimate the total bits from the error values or perform an exhaustive search. Making the estimation requires the total number of samples in the subframe, the number of overhead bits per order (by default, this is the number of bits per sample in the subframe, plus 5), and an error scale constant in addition to the LPC error values:

$$\text{Error Scale} = \frac{\ln(2)^2}{2 \times \text{Total Samples}} \tag{7.12}$$

Once the error scale has been calculated, one can generate a 'Bits per Residual' estimation function which, given an 'LPC Error' value, returns what its name implies:

$$\text{Bits per Residual(LPC Error)} = \frac{\ln(\text{Error Scale} \times \text{LPC Error})}{2 \times \ln(2)} \tag{7.13}$$

With this function, we can estimate how many bits the entire LPC subframe will take for each 'LPC Error' value and its associated 'Order':

$$\text{Total Bits(LPC Error, Order)} = \text{Bits per Residual(LPC Error)} \times (\text{Total Samples} - \text{Order}) + (\text{Order} \times \text{Overhead bits})$$

Continuing with our example, we have 20 samples and now have the error values of 2495, 2492 and 2490. This gives us an error scale of: $\frac{\ln(2)^2}{2\times 20} = \frac{.6931^2}{40} = .01201$

At LPC order 1, our bits per residual are:

$$\frac{\ln(.01201 \times 2495)}{2 \times \ln(2)} = \frac{\ln(29.96)}{1.386} = 2.453$$

And our total bits are:

$$(2.453 \times (20 - 1)) + (1 \times (16 + 5)) = 46.61 + 21 = 67.61$$

At LPC order 2, our bits per residual are:

$$\frac{\ln(.01201 \times 2492)}{2 \times \ln(2)} = \frac{\ln(29.92)}{1.386} = 2.452$$

And our total bits are:

$$(2.452 \times (20 - 2)) + (2 \times (16 + 5)) = 44.14 + 42 = 86.14$$

At LPC order 3, our bits per residual are:

$$\frac{\ln(.01201 \times 2490)}{2 \times \ln(2)} = \frac{\ln(29.90)}{1.386} = 2.451$$

And our total bits are:

$$(2.451 \times (20 - 3)) + (3 \times (16 + 5)) = 41.67 + 63 = 104.7$$

Therefore, since the total bits for order 1 are the smallest, the best order for this group of samples is 1.

Though as you'll notice, the bits per residual for order 3 were the smallest. So if this group of samples was very large, it's likely that order 3 would prevail since the residuals multiplied by a smaller bits per residual would counteract the relatively fixed overhead bits per order value.

Best Order Exhaustive Search

In a curious bit of recursion, finding the best order for an LPC subframe via an exhaustive search requires taking each list of LP Coefficients calculated previously, quantizing them into a list of QLP Coefficients and a QLP Shift Needed value,[3] determining the total amount of bits each hypothetical LPC subframe uses and using the LPC order which uses the fewest.

Remember that building an LPC subframe requires the following values: LPC Order, QLP Precision, QLP Shift Needed and QLP Coefficients along with the subframe's samples and bits-per-sample. For each possible LPC Order, the QLP Shift Needed and the QLP Coefficient list values can be calculated by quantizing the LP Coefficients. QLP Precision is the size of each QLP Coefficient list value in the subframe header. Simply choose the field with the largest number of bits in the QLP Coefficient list for the QLP Precision value.

Finally, instead of writing these hypothetical LPC subframes directly to disk, one only has to capture how many bits they *would* use. The hypothetical LPC subframe that uses the fewest number of bits is the one we should actually write to disk.

[3] Quantizing coefficients will be covered in the next section.

Quantizing Coefficients

Quantizing coefficients is a process of taking a list of LP Coefficients along with a QLP Coefficients Precision value and returning a list of QLP Coefficients and a QLP Shift Needed value. The first step is determining the upper and lower limits of the QLP Coefficients:

$$\text{QLP coefficient maximum} = 2^{precision-1} - 1 \tag{7.14}$$

$$\text{QLP coefficient minimum} = -2^{precision-1} \tag{7.15}$$

The QLP Coefficients Precision value is typically based on the encoder's block size:

Block Size	Precision	Block Size	Precision
Size $\leq$ 192	7	Size $\leq$ 384	8
Size $\leq$ 576	9	Size $\leq$ 1152	10
Size $\leq$ 2304	11	Size $\leq$ 4608	12
Size $>$ 4608	13		

So in our example of a block of 20 samples,

$$\text{QLP Coefficient maximum} = 2^{7-1} - 1 = 64 - 1 = 63$$
$$\text{QLP Coefficient minimum} = -2^{7-1} = -64$$

Now we determine the initial QLP Shift Needed value:

$$\text{shift} = \text{precision} - \left\lceil \frac{\log(\max(|\text{LP Coefficients}|))}{\log(2)} \right\rceil - 1 \tag{7.16}$$

where 'shift' is adjusted if necessary such that: $0 \leq \text{shift} \leq$ `0xF` , since it must fit into a 5-bit signed field and negative shifts are no-ops in the FLAC decoder.

Continuing our ongoing example, let's assume we're quantizing the LP coefficients 0.9065, -0.0052 and -0.0293. So our shift should be:

$$\text{shift} = 7 - \left\lceil \frac{\log(0.9065)}{\log(2)} \right\rceil - 1 = 7 - \left\lceil \frac{-0.0981}{0.6931} \right\rceil - 1 = 7 - 0 - 1 = 6$$

Finally, we determine the QLP Coefficient values themselves via a small recursive routine:

$$X(i) = E(i-1) + (\text{LP Coefficient}_i \times 2^{shift}) \tag{7.17}$$

$$\text{QLP Coefficient}_i = \text{round}(X(i)) \tag{7.18}$$

$$E(i) = X(i) - \text{QLP Coefficient}_i \tag{7.19}$$

where $E(0) = 0$ and each QLP Coefficient is adjusted prior to calculating the next $E(i)$ value such that: QLP coefficient minimum $\leq$ QLP Coefficient$_i$ $\leq$ QLP coefficient maximum

So to finish our LPC example:

$$\begin{aligned}
X(1) &= E(0) + (0.9065 \times 2^6) = 0 + 58.016 = \mathbf{58.016} \\
\text{QLP Coefficient}_1 &= \text{round}(58.016) = \mathbf{58} \\
E(1) &= X(1) - \text{QLP Coefficient}_1 = 58.016 - 58 = \mathbf{0.016} \\
X(2) &= E(1) + (-0.0052 \times 2^6) = 0.016 + -0.3328 = \mathbf{0.3168} \\
\text{QLP Coefficient}_2 &= \text{round}(0.3168) = \mathbf{0} \\
E(2) &= X(2) - \text{QLP Coefficient}_2 = 0.3168 - 0 = \mathbf{0.3168} \\
X(3) &= E(2) + (-0.0293 \times 2^6) = 0.3168 + -1.875 = \mathbf{-1.558} \\
\text{QLP Coefficient}_3 &= \text{round}(-1.558) = \mathbf{-2} \\
E(3) &= X(3) - \text{QLP Coefficient}_3 = -1.558 - -2 = \mathbf{0.4420}
\end{aligned}$$

Therefore, the LPC order is 3. The QLP Coefficients are 58, 0 and -2. The QLP Shift Needed value is 6. And, the QLP precision value can be calculated from the bits required for the largest absolute QLP Coefficient value. In this case, 6 bits are required to hold the value 58 so QLP precision can be 6.

Calculating LPC Residual

Warm-up samples and residuals are calculated as follows:

For $i = 0$ to $Order - 1$:

$$\text{Warm Up}_i = \text{Sample}_i$$

For $i = Order$ to Block Size - 1:

$$\text{Residual}_{i-Order} = \text{Sample}_i - \left\lfloor \frac{\sum_{j=0}^{Order-1} \text{QLP Coefficient}_j \times \text{Sample}_{i-j-1}}{2^{\text{QLP Shift Needed}}} \right\rfloor$$

Therefore, if 'LPC Order' is 3, 'QLP Shift Needed' is 6, and we have the following values:

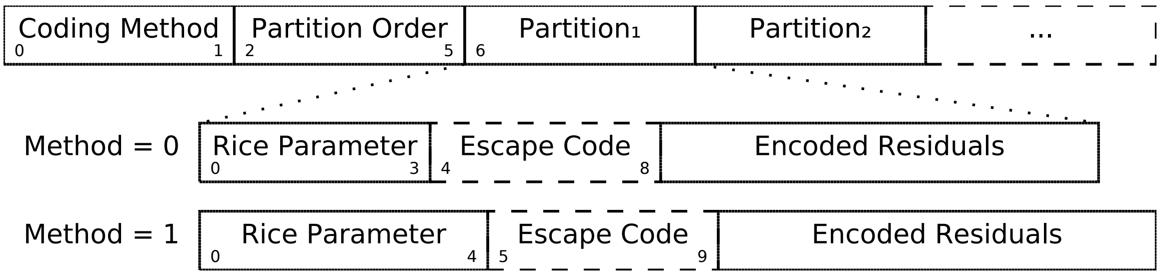
Sample_0	-37	LPC Coefficient_0	58
Sample_1	-33	LPC Coefficient_1	0
Sample_2	-36	LPC Coefficient_2	-2
Sample_3	-35		
Sample_4	-31		
Sample_5	-32		
Sample_6	-33		

$$
\begin{aligned}
\text{Warm Up}_0 &= \text{Sample}_0 = \textbf{-37} \\
\text{Warm Up}_1 &= \text{Sample}_1 = \textbf{-33} \\
\text{Warm Up}_2 &= \text{Sample}_2 = \textbf{-36} \\
\text{Residual}_0 &= \text{Sample}_3 - \left\lfloor \frac{(C_0 \times S_2) + (C_1 \times S_1) + (C_2 \times S_2)}{2^6} \right\rfloor \\
&= -35 - \left\lfloor \frac{(58 \times -36) + (0 \times -33) + (-2 \times -37)}{64} \right\rfloor \\
&= -35 - \left\lfloor \frac{-2014}{64} \right\rfloor = -35 - -32 = \textbf{-3} \\
\text{Residual}_1 &= \text{Sample}_4 - \left\lfloor \frac{(C_0 \times S_3) + (C_1 \times S_2) + (C_2 \times S_1)}{2^6} \right\rfloor \\
&= -31 - \left\lfloor \frac{(58 \times -35) + (0 \times -36) + (-2 \times -33)}{64} \right\rfloor \\
&= -31 - \left\lfloor \frac{-1964}{64} \right\rfloor = -31 - -31 = \textbf{0} \\
\text{Residual}_2 &= \text{Sample}_5 - \left\lfloor \frac{(C_0 \times S_4) + (C_1 \times S_3) + (C_2 \times S_2)}{2^6} \right\rfloor \\
&= -32 - \left\lfloor \frac{(58 \times -31) + (0 \times -35) + (-2 \times -36)}{64} \right\rfloor \\
&= -32 - \left\lfloor \frac{-1726}{64} \right\rfloor = -32 - -27 = \textbf{-5} \\
\text{Residual}_3 &= \text{Sample}_6 - \left\lfloor \frac{(C_0 \times S_5) + (C_1 \times S_4) + (C_2 \times S_3)}{2^6} \right\rfloor \\
&= -33 - \left\lfloor \frac{(58 \times -32) + (0 \times -31) + (-2 \times -35)}{64} \right\rfloor \\
&= -33 - \left\lfloor \frac{-1786}{64} \right\rfloor = -33 - -28 = \textbf{-5}
\end{aligned}
$$

Therefore, our warm-up samples are -37, -33 and -36; and our residual values are -3, 0, -5 and -5.

7.4.9 the Residual

Given a stream of residual values, one must place them in one or more partitions, each with its own Rice parameter, and prepended with a small header:



The residual's coding method is typically 0, unless one is encoding audio with more than 16 bits-per-sample and one of the partitions requests a Rice parameter higher than 2^4. The residual's partition order is chosen exhaustively, which means trying all of them within a certain range (e.g. 0 to 5) such that the residuals can be divided evenly between them and then the partition order which uses the smallest estimated amount of space is chosen.

Choosing the best Rice parameter is a matter of selecting the smallest value of 'x' such that:

$$\text{sample count} \times 2^x > \sum_{i=0}^{\text{residual count}-1} |\text{residual}_i| \tag{7.20}$$

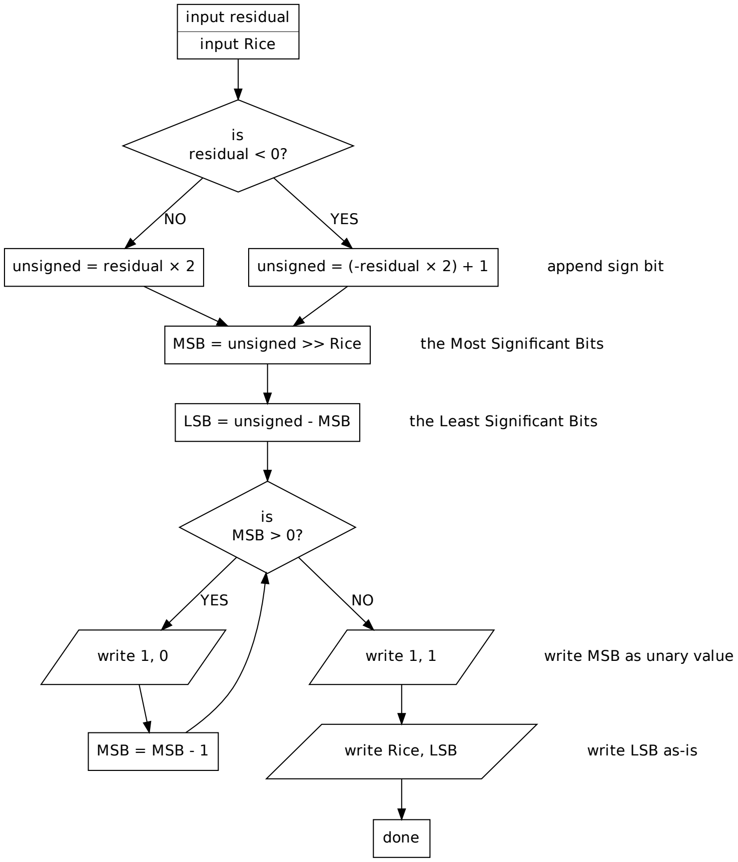
Again, this is easier to understand with a block of example residuals, 19 in total:
19×2^0 is not larger than 29.
19×2^1 is larger than 29, so the best Rice parameter for this block of residuals is 1.

Remember that the Rice parameter's maximum value is limited to 2^4 using coding method 0, or 2^5 using coding method 1.

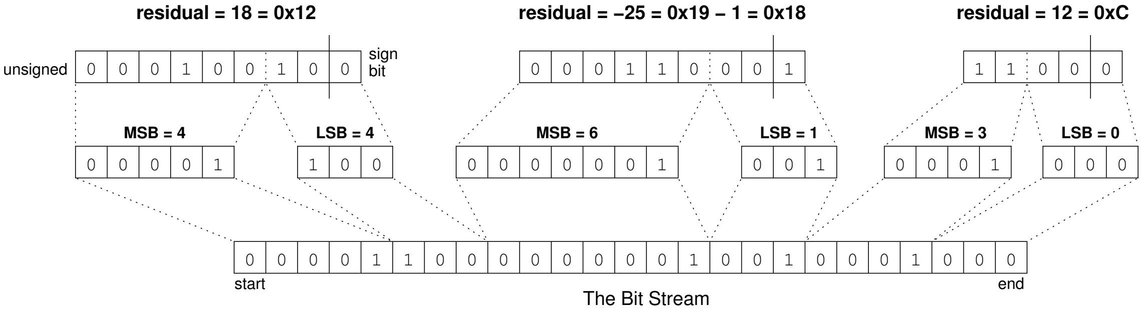
index	residual_i	$\lvert\text{residual}_i\rvert$
0	-1	1
1	1	1
2	1	1
3	1	1
4	0	0
5	3	3
6	0	0
7	-4	4
8	-1	1
9	0	0
10	1	1
11	1	1
12	1	1
13	4	4
14	-3	3
15	1	1
16	4	4
17	-1	1
18	-1	1
|sum|		**29**

Residual Values

Encoding individual residual values to Rice coding requires only the Rice parameter and the residuals themselves.



As an example, we'll encode the residual values 18, -25 and 12 with a Rice parameter of 3:



7.4.10 Checksums

Calculating the frame header's CRC-8 and frame footer's CRC-16 is necessary both for FLAC encoders and decoders, but the process is the same for each.

CRC-8

Given a byte of input and the previous CRC-8 checksum, or 0 as an initial value, the current checksum can be calculated as follows:

$$\text{checksum}_i = \text{CRC8}(\textit{byte} \textbf{ xor } \text{checksum}_{i-1}) \tag{7.21}$$

	0x?0	0x?1	0x?2	0x?3	0x?4	0x?5	0x?6	0x?7	0x?8	0x?9	0x?A	0x?B	0x?C	0x?D	0x?E	0x?F
0x0?	0x00	0x07	0x0E	0x09	0x1C	0x1B	0x12	0x15	0x38	0x3F	0x36	0x31	0x24	0x23	0x2A	0x2D
0x1?	0x70	0x77	0x7E	0x79	0x6C	0x6B	0x62	0x65	0x48	0x4F	0x46	0x41	0x54	0x53	0x5A	0x5D
0x2?	0xE0	0xE7	0xEE	0xE9	0xFC	0xFB	0xF2	0xF5	0xD8	0xDF	0xD6	0xD1	0xC4	0xC3	0xCA	0xCD
0x3?	0x90	0x97	0x9E	0x99	0x8C	0x8B	0x82	0x85	0xA8	0xAF	0xA6	0xA1	0xB4	0xB3	0xBA	0xBD
0x4?	0xC7	0xC0	0xC9	0xCE	0xDB	0xDC	0xD5	0xD2	0xFF	0xF8	0xF1	0xF6	0xE3	0xE4	0xED	0xEA
0x5?	0xB7	0xB0	0xB9	0xBE	0xAB	0xAC	0xA5	0xA2	0x8F	0x88	0x81	0x86	0x93	0x94	0x9D	0x9A
0x6?	0x27	0x20	0x29	0x2E	0x3B	0x3C	0x35	0x32	0x1F	0x18	0x11	0x16	0x03	0x04	0x0D	0x0A
0x7?	0x57	0x50	0x59	0x5E	0x4B	0x4C	0x45	0x42	0x6F	0x68	0x61	0x66	0x73	0x74	0x7D	0x7A
0x8?	0x89	0x8E	0x87	0x80	0x95	0x92	0x9B	0x9C	0xB1	0xB6	0xBF	0xB8	0xAD	0xAA	0xA3	0xA4
0x9?	0xF9	0xFE	0xF7	0xF0	0xE5	0xE2	0xEB	0xEC	0xC1	0xC6	0xCF	0xC8	0xDD	0xDA	0xD3	0xD4
0xA?	0x69	0x6E	0x67	0x60	0x75	0x72	0x7B	0x7C	0x51	0x56	0x5F	0x58	0x4D	0x4A	0x43	0x44
0xB?	0x19	0x1E	0x17	0x10	0x05	0x02	0x0B	0x0C	0x21	0x26	0x2F	0x28	0x3D	0x3A	0x33	0x34
0xC?	0x4E	0x49	0x40	0x47	0x52	0x55	0x5C	0x5B	0x76	0x71	0x78	0x7F	0x6A	0x6D	0x64	0x63
0xD?	0x3E	0x39	0x30	0x37	0x22	0x25	0x2C	0x2B	0x06	0x01	0x08	0x0F	0x1A	0x1D	0x14	0x13
0xE?	0xAE	0xA9	0xA0	0xA7	0xB2	0xB5	0xBC	0xBB	0x96	0x91	0x98	0x9F	0x8A	0x8D	0x84	0x83
0xF?	0xDE	0xD9	0xD0	0xD7	0xC2	0xC5	0xCC	0xCB	0xE6	0xE1	0xE8	0xEF	0xFA	0xFD	0xF4	0xF3

For example, given the header bytes: 0xFF, 0xF8, 0xCC, 0x1C, 0x00 and 0xC0:

$$\begin{aligned}
\text{checksum}_0 &= \text{CRC8}(\texttt{0xFF} \textbf{ xor } \texttt{0x00}) = \text{CRC8}(\texttt{0xFF}) = \texttt{0xF3} \\
\text{checksum}_1 &= \text{CRC8}(\texttt{0xF8} \textbf{ xor } \texttt{0xF3}) = \text{CRC8}(\texttt{0x0B}) = \texttt{0x31} \\
\text{checksum}_2 &= \text{CRC8}(\texttt{0xCC} \textbf{ xor } \texttt{0x31}) = \text{CRC8}(\texttt{0xFD}) = \texttt{0xFD} \\
\text{checksum}_3 &= \text{CRC8}(\texttt{0x1C} \textbf{ xor } \texttt{0xFD}) = \text{CRC8}(\texttt{0xE1}) = \texttt{0xA9} \\
\text{checksum}_4 &= \text{CRC8}(\texttt{0x00} \textbf{ xor } \texttt{0xA9}) = \text{CRC8}(\texttt{0xA9}) = \texttt{0x56} \\
\text{checksum}_5 &= \text{CRC8}(\texttt{0xC0} \textbf{ xor } \texttt{0x56}) = \text{CRC8}(\texttt{0x96}) = \texttt{0xEB}
\end{aligned}$$

Thus, the next byte after the header should be `0xEB`. Furthermore, when the checksum byte itself is run through the checksumming procedure:

$$\text{checksum}_6 = \text{CRC8}(\texttt{0xEB} \textbf{ xor } \texttt{0xEB}) = \text{CRC8}(\texttt{0x00}) = \texttt{0x00}$$

the result will always be 0. This is a handy way to verify a frame header's checksum since the checksum of the header's bytes along with the header's checksum itself will always result in 0.

CRC-16

CRC-16 is used to checksum the entire FLAC frame, including the header and any padding bits after the final subframe. Given a byte of input and the previous CRC-16 checksum, or 0 as an initial value, the current checksum can be calculated as follows:

$$\text{checksum}_i = \text{CRC16}(\mathit{byte}\ \textbf{xor}\ (\text{checksum}_{i-1} \gg 8))\ \textbf{xor}\ (\text{checksum}_{i-1} \ll 8) \tag{7.22}$$

and the checksum is always truncated to 16-bits.

	0x?0	0x?1	0x?2	0x?3	0x?4	0x?5	0x?6	0x?7	0x?8	0x?9	0x?A	0x?B	0x?C	0x?D	0x?E	0x?F
0x0?	0000	8005	800f	000a	801b	001e	0014	8011	8033	0036	003c	8039	0028	802d	8027	0022
0x1?	8063	0066	006c	8069	0078	807d	8077	0072	0050	8055	805f	005a	804b	004e	0044	8041
0x2?	80c3	00c6	00cc	80c9	00d8	80dd	80d7	00d2	00f0	80f5	80ff	00fa	80eb	00ee	00e4	80e1
0x3?	00a0	80a5	80af	00aa	80bb	00be	00b4	80b1	8093	0096	009c	8099	0088	808d	8087	0082
0x4?	8183	0186	018c	8189	0198	819d	8197	0192	01b0	81b5	81bf	01ba	81ab	01ae	01a4	81a1
0x5?	01e0	81e5	81ef	01ea	81fb	01fe	01f4	81f1	81d3	01d6	01dc	81d9	01c8	81cd	81c7	01c2
0x6?	0140	8145	814f	014a	815b	015e	0154	8151	8173	0176	017c	8179	0168	816d	8167	0162
0x7?	8123	0126	012c	8129	0138	813d	8137	0132	0110	8115	811f	011a	810b	010e	0104	8101
0x8?	8303	0306	030c	8309	0318	831d	8317	0312	0330	8335	833f	033a	832b	032e	0324	8321
0x9?	0360	8365	836f	036a	837b	037e	0374	8371	8353	0356	035c	8359	0348	834d	8347	0342
0xA?	03c0	83c5	83cf	03ca	83db	03de	03d4	83d1	83f3	03f6	03fc	83f9	03e8	83ed	83e7	03e2
0xB?	83a3	03a6	03ac	83a9	03b8	83bd	83b7	03b2	0390	8395	839f	039a	838b	038e	0384	8381
0xC?	0280	8285	828f	028a	829b	029e	0294	8291	82b3	02b6	02bc	82b9	02a8	82ad	82a7	02a2
0xD?	82e3	02e6	02ec	82e9	02f8	82fd	82f7	02f2	02d0	82d5	82df	02da	82cb	02ce	02c4	82c1
0xE?	8243	0246	024c	8249	0258	825d	8257	0252	0270	8275	827f	027a	826b	026e	0264	8261
0xF?	0220	8225	822f	022a	823b	023e	0234	8231	8213	0216	021c	8219	0208	820d	8207	0202

For example, given the frame bytes: 0xFF, 0xF8, 0xCC, 0x1C, 0x00, 0xC0, 0xEB, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00, 0x00 and 0x00, the frame's CRC-16 can be calculated as follows:

$$\begin{aligned}
\text{checksum}_0 &= \text{CRC16}(\texttt{0xFF}\ \textbf{xor}\ (\texttt{0x0000} \gg 8))\ \textbf{xor}\ (\texttt{0x0000} \ll 8) = \text{CRC16}(\texttt{0xFF})\ \textbf{xor}\ \texttt{0x0000} = \texttt{0x0202}\\
\text{checksum}_1 &= \text{CRC16}(\texttt{0xF8}\ \textbf{xor}\ (\texttt{0x0202} \gg 8))\ \textbf{xor}\ (\texttt{0x0202} \ll 8) = \text{CRC16}(\texttt{0xFA})\ \textbf{xor}\ \texttt{0x0200} = \texttt{0x001C}\\
\text{checksum}_2 &= \text{CRC16}(\texttt{0xCC}\ \textbf{xor}\ (\texttt{0x001C} \gg 8))\ \textbf{xor}\ (\texttt{0x001C} \ll 8) = \text{CRC16}(\texttt{0xCC})\ \textbf{xor}\ \texttt{0x1C00} = \texttt{0x1EA8}\\
\text{checksum}_3 &= \text{CRC16}(\texttt{0x1C}\ \textbf{xor}\ (\texttt{0x1EA8} \gg 8))\ \textbf{xor}\ (\texttt{0x1EA8} \ll 8) = \text{CRC16}(\texttt{0x02})\ \textbf{xor}\ \texttt{0xA800} = \texttt{0x280F}\\
\text{checksum}_4 &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x280F} \gg 8))\ \textbf{xor}\ (\texttt{0x280F} \ll 8) = \text{CRC16}(\texttt{0x28})\ \textbf{xor}\ \texttt{0x0F00} = \texttt{0x0FF0}\\
\text{checksum}_5 &= \text{CRC16}(\texttt{0xC0}\ \textbf{xor}\ (\texttt{0x0FF0} \gg 8))\ \textbf{xor}\ (\texttt{0x0FF0} \ll 8) = \text{CRC16}(\texttt{0xCF})\ \textbf{xor}\ \texttt{0xF000} = \texttt{0xF2A2}\\
\text{checksum}_6 &= \text{CRC16}(\texttt{0xEB}\ \textbf{xor}\ (\texttt{0xF2A2} \gg 8))\ \textbf{xor}\ (\texttt{0xF2A2} \ll 8) = \text{CRC16}(\texttt{0x19})\ \textbf{xor}\ \texttt{0xA200} = \texttt{0x2255}\\
\text{checksum}_7 &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x2255} \gg 8))\ \textbf{xor}\ (\texttt{0x2255} \ll 8) = \text{CRC16}(\texttt{0x22})\ \textbf{xor}\ \texttt{0x5500} = \texttt{0x55CC}\\
\text{checksum}_8 &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x55CC} \gg 8))\ \textbf{xor}\ (\texttt{0x55CC} \ll 8) = \text{CRC16}(\texttt{0x55})\ \textbf{xor}\ \texttt{0xCC00} = \texttt{0xCDFE}\\
\text{checksum}_9 &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0xCDFE} \gg 8))\ \textbf{xor}\ (\texttt{0xCDFE} \ll 8) = \text{CRC16}(\texttt{0xCD})\ \textbf{xor}\ \texttt{0xFE00} = \texttt{0x7CAD}\\
\text{checksum}_{10} &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x7CAD} \gg 8))\ \textbf{xor}\ (\texttt{0x7CAD} \ll 8) = \text{CRC16}(\texttt{0x7C})\ \textbf{xor}\ \texttt{0xAD00} = \texttt{0x2C0B}\\
\text{checksum}_{11} &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x2C0B} \gg 8))\ \textbf{xor}\ (\texttt{0x2C0B} \ll 8) = \text{CRC16}(\texttt{0x2C})\ \textbf{xor}\ \texttt{0x0B00} = \texttt{0x8BEB}\\
\text{checksum}_{12} &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x8BEB} \gg 8))\ \textbf{xor}\ (\texttt{0x8BEB} \ll 8) = \text{CRC16}(\texttt{0x8B})\ \textbf{xor}\ \texttt{0xEB00} = \texttt{0xE83A}\\
\text{checksum}_{13} &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0xE83A} \gg 8))\ \textbf{xor}\ (\texttt{0xE83A} \ll 8) = \text{CRC16}(\texttt{0xE8})\ \textbf{xor}\ \texttt{0x3A00} = \texttt{0x3870}\\
\text{checksum}_{14} &= \text{CRC16}(\texttt{0x00}\ \textbf{xor}\ (\texttt{0x3870} \gg 8))\ \textbf{xor}\ (\texttt{0x3870} \ll 8) = \text{CRC16}(\texttt{0x38})\ \textbf{xor}\ \texttt{0x7000} = \texttt{0xF093}
\end{aligned}$$

Thus, the next two bytes after the final subframe should be `0xF0` and `0x93`. Again, when the checksum bytes are run through the checksumming procedure:

$$\begin{aligned}
\text{checksum}_{15} &= \text{CRC16}(\texttt{0xF0}\ \textbf{xor}\ (\texttt{0xF093} \gg 8))\ \textbf{xor}\ (\texttt{0xF093} \ll 8) = \text{CRC16}(\texttt{0x00})\ \textbf{xor}\ \texttt{0x9300} = \texttt{0x9300}\\
\text{checksum}_{16} &= \text{CRC16}(\texttt{0x93}\ \textbf{xor}\ (\texttt{0x9300} \gg 8))\ \textbf{xor}\ (\texttt{0x9300} \ll 8) = \text{CRC16}(\texttt{0x00})\ \textbf{xor}\ \texttt{0x0000} = \texttt{0x0000}
\end{aligned}$$

the result will also always be 0, just as in the CRC-8.

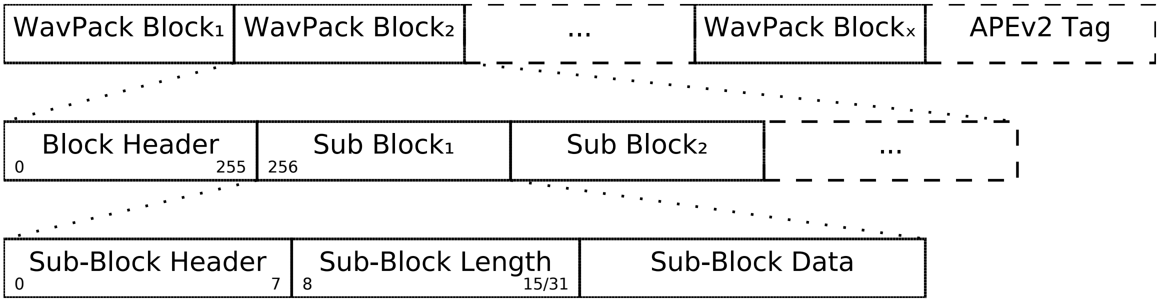
8 WavPack

WavPack is a format for compressing Wave files, typically in lossless mode. Notably, it also has a lossy mode and even a hybrid mode which allows the ‘correction’ file to be separated from a lossy core.

Metadata is stored as an APEv2 tag, which is described on page 112.

Its stream of data is stored little-endian, as described on page 17.

8.1 the WavPack File Stream



8.2 the WavPack Block Header

Block ID `wvpk' (0x6B707677) 0 31		Block Size 32 63	
Version 64 79	Track Number 80 87	Index Number 88 95	Total Samples 96 127
Block Index 128 159		Block Samples 160 191	
Bits per Sample 192 193	Mono Output 194	Hybrid Mode 195	Joint Stereo 196
Channel Decorrelation 197	Hbd. Noise Shaping 198	Floating Point Data 199	Extended Size Integers 200
Hbd. Controls Bitrate 201	Hbd. Noise Balanced 202	Initial Block 203	Final Block 204
Left Shift Data 205 209	Maximum Magnitude 210 214	Sample Rate 215 218	
Reserved 219 220	Use IIR 221	False Stereo 222	Reserved 223
CRC 224 255			

'Block Size' is the length of everything in the block past the 'Block Size' field itself - or everything in the block past the CRC, minus 24 bytes.

'Bits per Sample' is one of 4 values:

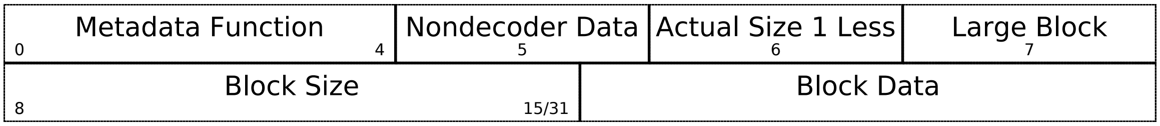
00 = 8 bps, 01 = 16 bps, 10 = 24 bps, 11 = 32 bps .

'Mono Output' bit indicates the channel count. If 1, this block has 1 channel. If 0, this block has 2 channels. For an audio stream with more than 2 channels, check the 'Initial Block' and 'Final Block' bits to indicate the start and end of the channels. As an example:

Initial Block	Final Block	Mono Output	Channels
1	0	0	2
0	0	1	1
0	0	1	1
0	1	0	2
		Total	6

value	sample rate
0000	6000
0001	8000
0010	9600
0011	11025
0100	12000
0101	16000
0110	22050
0111	24000
1000	32000
1001	44100
1010	48000
1011	64000
1100	88200
1101	96000
1110	192000
1111	reserved

8.2.1 WavPack Sub-Block



If the ‘Large Block’ field is 0, the ‘Block Size’ field is 8 bits long. If it is 1, the ‘Block Size’ field is 24 bits long. The ‘Block Size’ field is the length of ‘Block Data’, in 16-bit words rather than bytes. If ‘Actual Size 1 Less’ is set, that means ‘Block Data’ doesn’t contain an even number of bytes; it is padded with a single null byte at the end in order to fit. If ‘Nondecoder Data’ is set, that means the decoder does not have to understand the contents of this particular sub-block in order to decode the audio.

8.3 WavPack Decoding

Decoding each WavPack block requires reading its sub-blocks as ‘arguments’ to the decoder. One can envision them like named arguments to a function call since many sub-blocks may be optional or appear in an arbitrary order. As a sort of hypothetical high-level example:

```
decode_block(decorrelation_terms=sub_block[0],
             decorrelation_weights=sub_block[1],
             decorrelation_samples=sub_block[2],
             entropy_variables=sub_block[3],
             bitstream=sub_block[4])
```

Every block containing audio data requires ‘Entropy Variables’ and ‘Bitstream’ sub-blocks. The ‘Decorrelation Terms’, ‘Decorrelation Weights’ and ‘Decorrelation Samples’ sub-blocks are for performing one or more decorrelation passes over the bitstream’s samples.

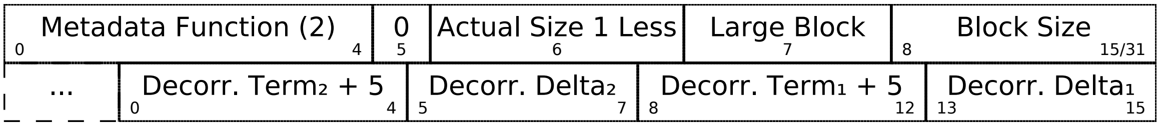
Each block will decode to 1 or 2 channels of raw PCM output. Since files may have more than 2 channels, we may need to decode several blocks in order to retrieve all the channels of data so they can be properly combined.

8.3.1 False Stereo

If the 'False Stereo' bit is set in the block header, treat the block as mono for decoding purposes until just before the channel's data is output.

8.3.2 the Decorrelation Terms Sub-Block

This block contains the decorrelation terms and deltas values. The quantity of those values indicates how many decorrelation passes we'll be performing over the bitstream sub-block's samples. One can presume this sub-block will occur prior to the 'Decorrelation Weights' and 'Decorrelation Samples' sub-blocks.



The number of decorrelation terms and deltas equals the 'Block Size' times 2, and minus 1 if 'Actual Size 1 Less' is set. Each term and delta pair is 8 bits and stored in *reverse* order. In addition, one must subtract 5 from the stored value of each unsigned term to get its actual value.

For example, given the complete sub-block bytes:

```
42 03 57 57 47 56 48 00
```

we have a total of 5 term/delta pairs whose values are as follows:

Decorrelation Term$_5$	0x17 - 5 = 18	Decorrelation Delta$_5$	0x2 = 2
Decorrelation Term$_4$	0x17 - 5 = 18	Decorrelation Delta$_4$	0x2 = 2
Decorrelation Term$_3$	0x07 - 5 = 2	Decorrelation Delta$_3$	0x2 = 2
Decorrelation Term$_2$	0x16 - 5 = 17	Decorrelation Delta$_2$	0x2 = 2
Decorrelation Term$_1$	0x08 - 5 = 3	Decorrelation Delta$_1$	0x2 = 2

Remember that this is a little-endian stream and that the least-significant bits (the 'Decorrelation Delta' value) are on the left side of each byte.

8.3.3 the Decorrelation Weights Sub-Block

Metadata Function (3)	0	Actual Size 1 Less	Large Block	Block Size
0 – 4	5	6	7	8 – 15/31

...	Decorrelation Weight₂	Decorrelation Weight₁
	0 – 7	8 – 15

As with the decorrelations terms sub-block, the decorrelation weights are stored in reverse order. The number of weights stored can be determined from the sub-block's size. Each is stored in a signed, 8-bit field and interleaved between channels in the case of 2 channel blocks. For example, Decorrelation Weight$_1$ is for channel 'B', Decorrelation Weight$_2$ is for channel 'A'[1], Decorrelation Weight$_3$ is for channel 'B' and so on such that the first weight value in the sub-block will be for the highest 'Decorrelation Weight A'. Converting the 8-bit values to the actual decorrelation weights requires the following formula:

$$\text{Decorrelation Weight} = \begin{cases} \text{value} \times 2^3 + \left\lfloor \frac{\text{value} \times 2^3 + 2^6}{2^7} \right\rfloor & \text{if value} > 0 \\ 0 & \text{if value} = 0 \\ \text{value} \times 2^3 & \text{if value} < 0 \end{cases}$$

For example, given a 2 channel block with 5 decorrelation terms and the sub-block bytes:

```
03 05 06 06 06 06 04 04 06 06 02 03
```

our 'Decorrelation Weights' are as follows:

Weight A_5	$6 \times 2^3 + \left\lfloor \frac{6 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 48$	Weight B_5	$6 \times 2^3 + \left\lfloor \frac{6 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 48$
Weight A_4	$6 \times 2^3 + \left\lfloor \frac{6 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 48$	Weight B_4	$6 \times 2^3 + \left\lfloor \frac{6 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 48$
Weight A_3	$4 \times 2^3 + \left\lfloor \frac{4 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 32$	Weight B_3	$4 \times 2^3 + \left\lfloor \frac{4 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 32$
Weight A_2	$6 \times 2^3 + \left\lfloor \frac{6 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 48$	Weight B_2	$6 \times 2^3 + \left\lfloor \frac{6 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 48$
Weight A_1	$2 \times 2^3 + \left\lfloor \frac{2 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 16$	Weight B_1	$3 \times 2^3 + \left\lfloor \frac{3 \times 2^3 + 2^6}{2^7} \right\rfloor$	$= 24$

Note that decoding a WavPack file requires having the same number of 'Decorrelation Weight' values, per channel, as 'Decorrelation Terms' values. However, this block may contain less. In that event, those low weight values are set to 0.

[1] Why 'A' and 'B'? Since a block may be only two channels out of many, it makes sense not to number them to avoid ambiguity.

8.3.4 the Decorrelation Samples Sub-Block

Metadata Function (4)	0	Actual Size 1 Less	Large Block	Block Size
0 4	5	6	7	8 15/31

Decorrelation Sample₁	Decorrelation Sample₂	...
0 15	16 31	

The decorrelation samples values are stored as signed, 16-bit values. Converting them to sample values requires the following formula:

$$\text{Sample} = \begin{cases} \lfloor \text{wv\_exp2}(value \bmod 256) \div 2^{9-\lfloor value \div 2^8 \rfloor} \rfloor & \text{if } 0 \leq value \leq 2304 \\ \text{wv\_exp2}(value \bmod 256) \times 2^{\lfloor value \div 2^8 \rfloor - 9} & \text{if } 2304 < value \leq 32767 \\ -\lfloor \text{wv\_exp2}(-value \bmod 256) \div 2^{9-\lfloor -value \div 2^8 \rfloor} \rfloor & \text{if } -2304 \leq value < 0 \\ -(\text{wv\_exp2}(-value \bmod 256) \times 2^{\lfloor -value \div 2^8 \rfloor - 9}) & \text{if } -32768 \leq value < -2304 \end{cases}$$

where 'wv_exp2' is defined from the following base-16 table:

	0x?0	0x?1	0x?2	0x?3	0x?4	0x?5	0x?6	0x?7	0x?8	0x?9	0x?A	0x?B	0x?C	0x?D	0x?E	0x?F
0x0?	100	101	101	102	103	103	104	105	106	106	107	108	108	109	10A	10B
0x1?	10B	10C	10D	10E	10E	10F	110	110	111	112	113	113	114	115	116	116
0x2?	117	118	119	119	11A	11B	11C	11D	11D	11E	11F	120	120	121	122	123
0x3?	124	124	125	126	127	128	128	129	12A	12B	12C	12C	12D	12E	12F	130
0x4?	130	131	132	133	134	135	135	136	137	138	139	13A	13A	13B	13C	13D
0x5?	13E	13F	140	141	141	142	143	144	145	146	147	148	148	149	14A	14B
0x6?	14C	14D	14E	14F	150	151	151	152	153	154	155	156	157	158	159	15A
0x7?	15B	15C	15D	15E	15E	15F	160	161	162	163	164	165	166	167	168	169
0x8?	16A	16B	16C	16D	16E	16F	170	171	172	173	174	175	176	177	178	179
0x9?	17A	17B	17C	17D	17E	17F	180	181	182	183	184	185	187	188	189	18A
0xA?	18B	18C	18D	18E	18F	190	191	192	193	195	196	197	198	199	19A	19B
0xB?	19C	19D	19F	1A0	1A1	1A2	1A3	1A4	1A5	1A6	1A8	1A9	1AA	1AB	1AC	1AD
0xC?	1AF	1B0	1B1	1B2	1B3	1B4	1B6	1B7	1B8	1B9	1BA	1BC	1BD	1BE	1BF	1C0
0xD?	1C2	1C3	1C4	1C5	1C6	1C8	1C9	1CA	1CB	1CD	1CE	1CF	1D0	1D2	1D3	1D4
0xE?	1D6	1D7	1D8	1D9	1DB	1DC	1DD	1DE	1E0	1E1	1E2	1E4	1E5	1E6	1E8	1E9
0xF?	1EA	1EC	1ED	1EE	1F0	1F1	1F2	1F4	1F5	1F6	1F8	1F9	1FA	1FC	1FD	1FF

For example, given the sub-frame bytes:

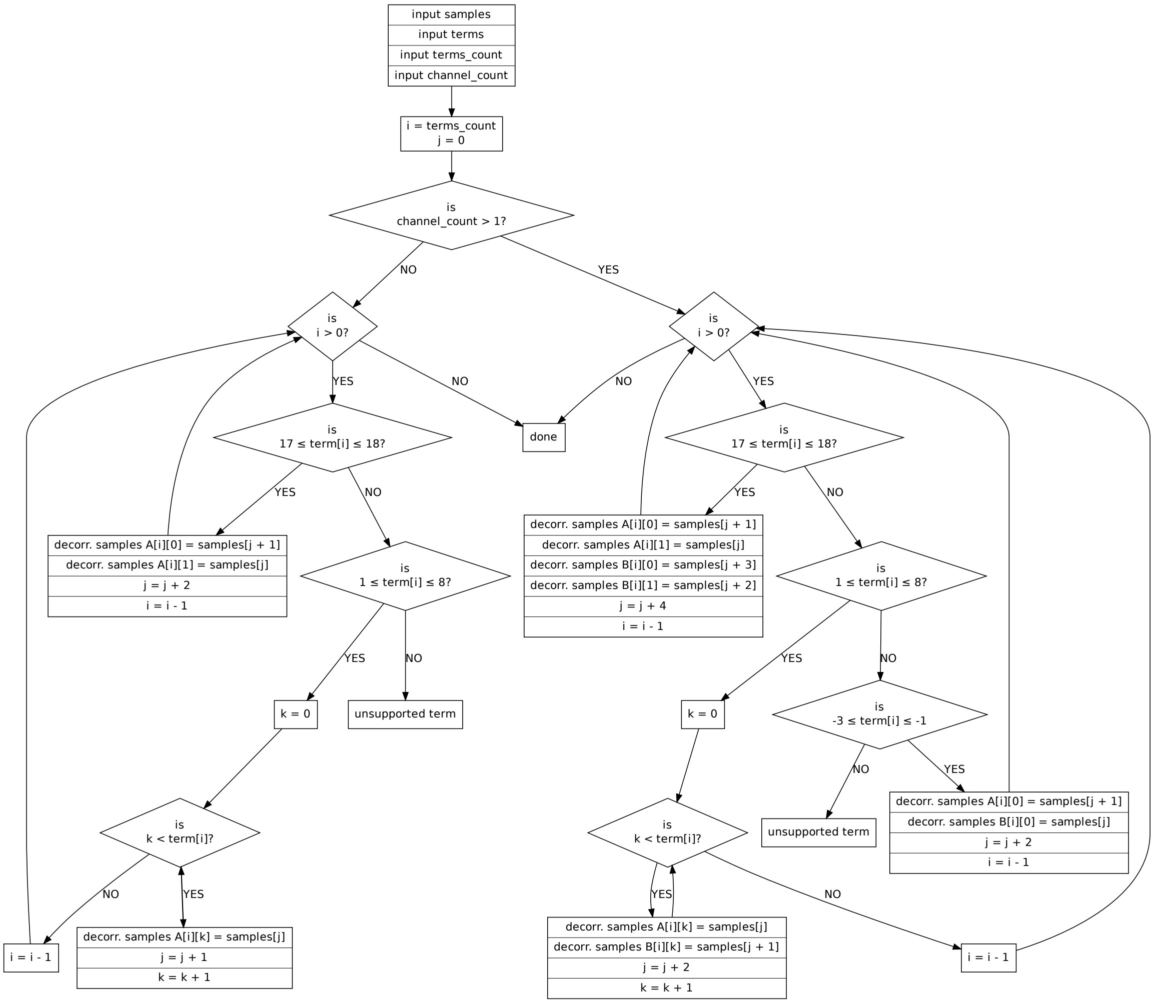
```
04 04 CF F8 B7 F8 CF 05 B3 05
```

our 'Decorrelation Sample' values are:

$$\begin{aligned} \text{Sample}_1 &= \texttt{0xF8CF} = -1841 = -\lfloor \text{wv\_exp2}(1841 \bmod 256) \div 2^{9-\lfloor 1841 \div 2^8 \rfloor} \rfloor \\ &= -\lfloor \text{wv\_exp2}(49) \div 2^{9-7} \rfloor = -\lfloor 292 \div 4 \rfloor = \mathbf{-73} \\ \text{Sample}_2 &= \texttt{0xF8B7} = -1865 = -\lfloor \text{wv\_exp2}(1865 \bmod 256) \div 2^{9-\lfloor 1865 \div 2^8 \rfloor} \rfloor \\ &= -\lfloor \text{wv\_exp2}(73) \div 2^{9-7} \rfloor = -\lfloor 312 \div 4 \rfloor = \mathbf{-78} \\ \text{Sample}_3 &= \texttt{0x05CF} = 1487 = \lfloor \text{wv\_exp2}(1487 \bmod 256) \div 2^{9-\lfloor 1487 \div 2^8 \rfloor} \rfloor \\ &= \lfloor \text{wv\_exp2}(207) \div 2^{9-5} \rfloor = \lfloor 448 \div 16 \rfloor = \mathbf{28} \\ \text{Sample}_4 &= \texttt{0x05B3} = 1459 = \lfloor \text{wv\_exp2}(1459 \bmod 256) \div 2^{9-\lfloor 1459 \div 2^8 \rfloor} \rfloor \\ &= \lfloor \text{wv\_exp2}(179) \div 2^{9-5} \rfloor = \lfloor 416 \div 16 \rfloor = \mathbf{26} \end{aligned}$$

We're not done yet, however. The next step is to determine which set of 'Decorrelation Sample' values correspond to which 'Decorrelation Term'[2].

As with 'Decorrelation Weights', 'Decorrelation Sample' values are stored in reverse order, alternate between channels and depend on the corresponding 'Decorrelation Term' values.

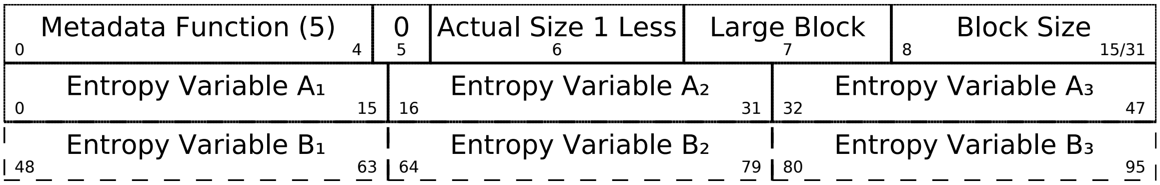


It's likely that the decorrelation sample assignment process will request more samples than this sub-block contains. In that event, treat those samples as 0.

[2] As extracted on page 74

8.3.5 the Entropy Variables Sub-Block

Whereas the three preceding sub-blocks are for performing decorrelation passes, this sub-block is required for decoding the 'Bitstream' sub-block's data. These entropy variables are median values which are stored as fractions of integers.



If a block is mono, this sub-block contains 3 'Entropy Variables'. If a block is stereo, this sub-block contains 6. Each is stored as a signed, 16-bit value which is packed in the same fashion as 'Decorrelation Samples'[3]. For example, given a 2 channel block with the sub-block bytes:

```
05 06 e2 07 9b 08 55 09 e2 07 76 08 ba 08
```

our 'Entropy Variables' are:

$$\begin{aligned}
\text{Entropy Variable A}_1 &= \texttt{0x07E2} = 2018 = \lfloor \text{wv\_exp2}(2018 \bmod 256) \div 2^{9-\lfloor 2018 \div 2^8 \rfloor} \rfloor \\
&= \lfloor \text{wv\_exp2}(226) \div 2^{9-7} \rfloor = \lfloor 472 \div 4 \rfloor = \mathbf{118} \\
\text{Entropy Variable A}_2 &= \texttt{0x089B} = 2203 = \lfloor \text{wv\_exp2}(2203 \bmod 256) \div 2^{9-\lfloor 2203 \div 2^8 \rfloor} \rfloor \\
&= \lfloor \text{wv\_exp2}(155) \div 2^{9-8} \rfloor = \lfloor 389 \div 2 \rfloor = \mathbf{194} \\
\text{Entropy Variable A}_3 &= \texttt{0x0955} = 2389 = \text{wv\_exp2}(2389 \bmod 256) \times 2^{\lfloor 2389 \div 2^8 \rfloor - 9} \\
&= \text{wv\_exp2}(85) \times 2^{9-9} = 322 \times 1 = \mathbf{322} \\
\text{Entropy Variable B}_1 &= \texttt{0x07E2} = 2018 = \lfloor \text{wv\_exp2}(2018 \bmod 256) \div 2^{9-\lfloor 2018 \div 2^8 \rfloor} \rfloor \\
&= \lfloor \text{wv\_exp2}(226) \div 2^{9-7} \rfloor = \lfloor 472 \div 4 \rfloor = \mathbf{118} \\
\text{Entropy Variable B}_2 &= \texttt{0x0876} = 2166 = \lfloor \text{wv\_exp2}(2166 \bmod 256) \div 2^{9-\lfloor 2166 \div 2^8 \rfloor} \rfloor \\
&= \lfloor \text{wv\_exp2}(118) \div 2^{9-8} \rfloor = \lfloor 352 \div 2 \rfloor = \mathbf{176} \\
\text{Entropy Variable B}_3 &= \texttt{0x08BA} = 2234 = \lfloor \text{wv\_exp2}(2234 \bmod 256) \div 2^{9-\lfloor 2234 \div 2^8 \rfloor} \rfloor \\
&= \lfloor \text{wv\_exp2}(186) \div 2^{9-8} \rfloor = \lfloor 424 \div 2 \rfloor = \mathbf{212}
\end{aligned}$$

[3] As described on page 76.

8.3.6 the Bitstream Sub-Block

This sub-block contains the block's residual values. Once decoded, these will ultimately become our file's raw PCM values.

Metadata Function (0xA) 0 4	0 5	Actual Size 1 Less 6	Large Block 7	Block Size 8 15/31
Residual Data 32				

Decoding the 'Bitstream' sub-block requires the 'Entropy Variables' data which it combines with this sub-block's bitstream to yield a set of signed values totaling 'Block Samples' (times 2 if the block is stereo).

Decoding each value is a complex process which I'll divide into three separate steps.

As an example, we'll use a 1-channel block with the entropy variables:

- Entropy Variable $A_1 = 111$
- Entropy Variable $A_1 = 159$
- Entropy Variable $A_1 = 299$

and the partial bitstream sub-block bytes:

```
8a 0e 00 00 a1 77 e9
```

The first four bytes are the header and block size values. The remaining three are as follows as a little-endian stream:

```
1 0 0 0 0 1 0 1  1 1 1 0 1 1 1 0  1 0 0 1 0 1 1 1
```

Determining t

The first step is taking two boolean values called 'Holding One' and 'Holding Zero' and determining 't'. These holding values can be thought of as registers in a sort of WavPack bitstream virtual machine whose values will change over the course of decoding. Their initial values are both false.

'limited unary' means counting the number of `1` bits until the next `0` bit, to a maximum of 33, `1` bits in a row.

In our example, to decode the first 't' value:

- **is holding_zero?** no
- **t = limited_unary** = '`1 0`' = 1
- **is t = 16?** no
- **is holding_one?** no
- **is t odd?** yes
- **holding_one = true**
- **holding_zero = false**
- **t = $\lfloor$ t $\div$ 2 $\rfloor$** = $\lfloor 1 \div 2 \rfloor = 0$

So our 't' value is 0, our 'Holding One' value is true, our 'Holding Zero' value is false and we've consumed 2 bits from the sub-block's bitstream.

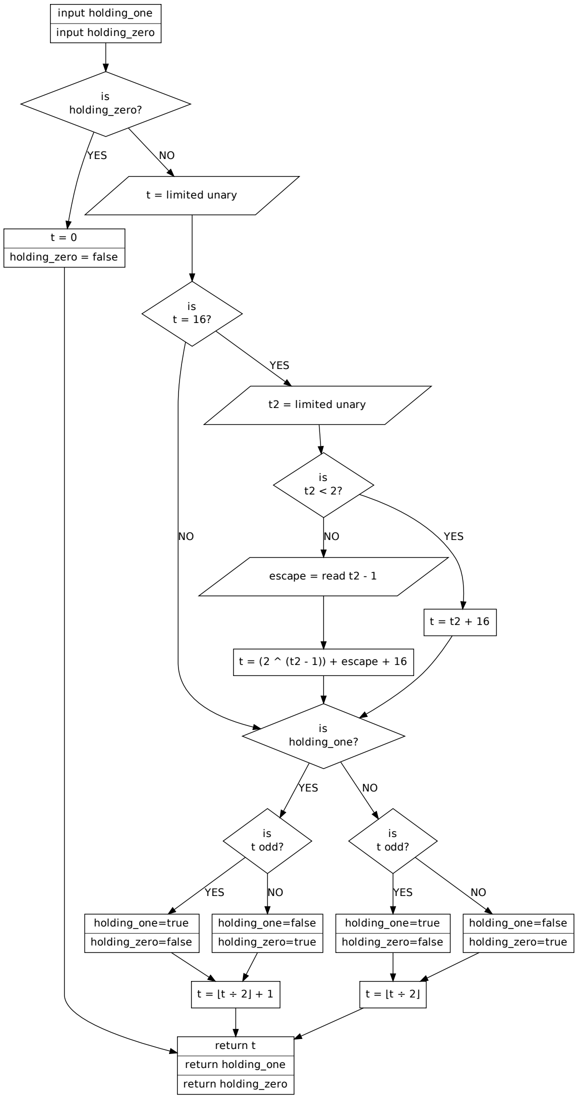


Figure 8.1: Step 1: determining t

Calculating Base/Add

The next step is taking 't' and calculating 'Base' and 'Add' from our entropy variables, updating our entropy variables in the process.

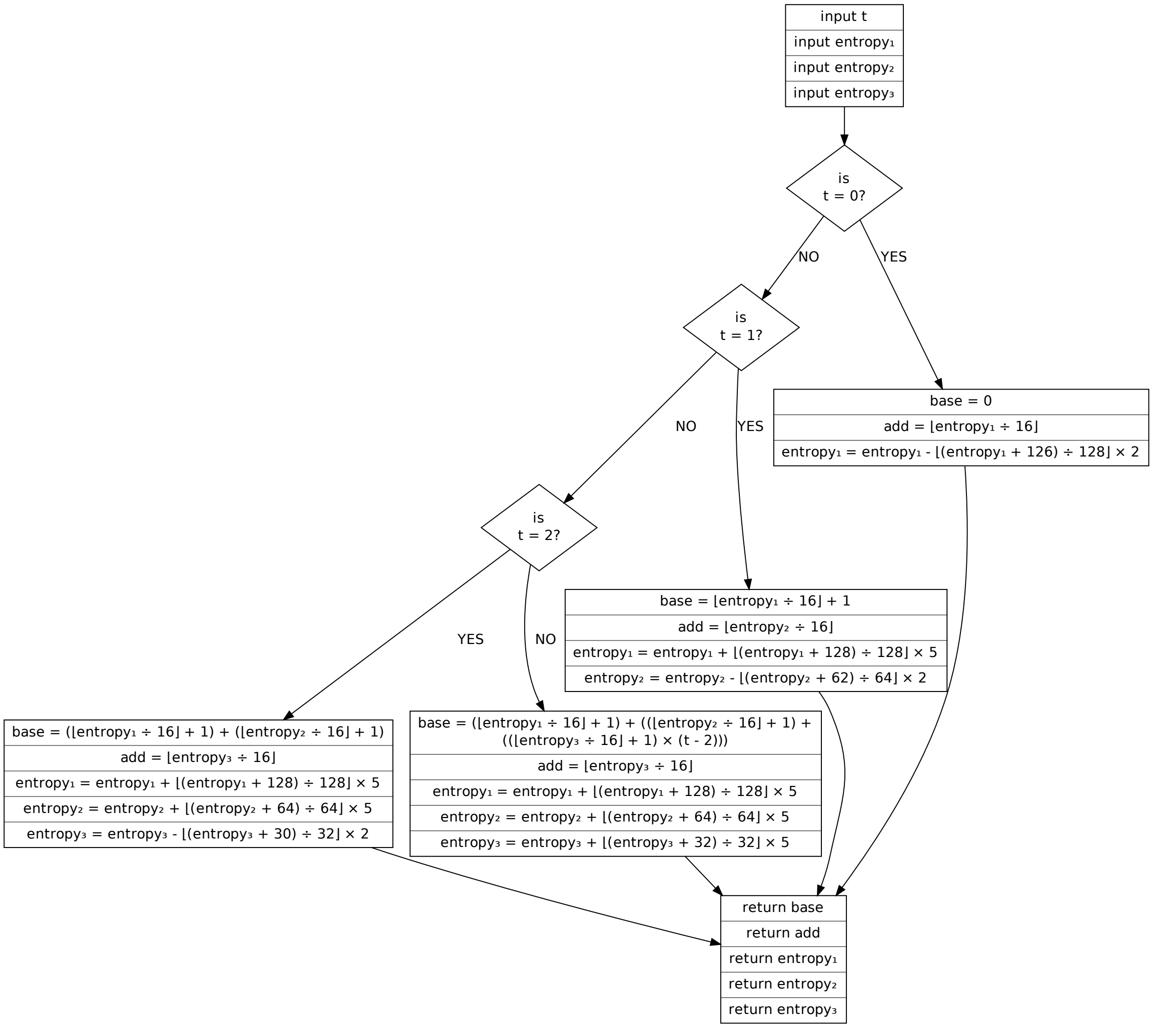

input t
input entropy_1
input entropy_2
input entropy_3
is t = 0?
NO
YES
is t = 1?
base = 0
add = $\lfloor \text{entropy}_1 \div 16 \rfloor$
$\text{entropy}_1 = \text{entropy}_1 - \lfloor (\text{entropy}_1 + 126) \div 128 \rfloor \times 2$
NO
YES
is t = 2?
base = $\lfloor \text{entropy}_1 \div 16 \rfloor + 1$
add = $\lfloor \text{entropy}_2 \div 16 \rfloor$
$\text{entropy}_1 = \text{entropy}_1 + \lfloor (\text{entropy}_1 + 128) \div 128 \rfloor \times 5$
$\text{entropy}_2 = \text{entropy}_2 - \lfloor (\text{entropy}_2 + 62) \div 64 \rfloor \times 2$
YES
NO
base = $(\lfloor \text{entropy}_1 \div 16 \rfloor + 1) + (\lfloor \text{entropy}_2 \div 16 \rfloor + 1)$
add = $\lfloor \text{entropy}_3 \div 16 \rfloor$
$\text{entropy}_1 = \text{entropy}_1 + \lfloor (\text{entropy}_1 + 128) \div 128 \rfloor \times 5$
$\text{entropy}_2 = \text{entropy}_2 + \lfloor (\text{entropy}_2 + 64) \div 64 \rfloor \times 5$
$\text{entropy}_3 = \text{entropy}_3 - \lfloor (\text{entropy}_3 + 30) \div 32 \rfloor \times 2$
base = $(\lfloor \text{entropy}_1 \div 16 \rfloor + 1) + ((\lfloor \text{entropy}_2 \div 16 \rfloor + 1) + ((\lfloor \text{entropy}_3 \div 16 \rfloor + 1) \times (t - 2)))$
add = $\lfloor \text{entropy}_3 \div 16 \rfloor$
$\text{entropy}_1 = \text{entropy}_1 + \lfloor (\text{entropy}_1 + 128) \div 128 \rfloor \times 5$
$\text{entropy}_2 = \text{entropy}_2 + \lfloor (\text{entropy}_2 + 64) \div 64 \rfloor \times 5$
$\text{entropy}_3 = \text{entropy}_3 + \lfloor (\text{entropy}_3 + 32) \div 32 \rfloor \times 5$
return base
return add
return entropy_1
return entropy_2
return entropy_3

Figure 8.2: Step 2: determining base/add

So to continue our example:

- **is t = 0?** yes
- **base** $= 0$
- **add =** $\lfloor \textbf{Entropy}_1 \div 16 \rfloor = \lfloor 111 \div 16 \rfloor = 6$
- $\textbf{Entropy}_1 = \textbf{Entropy}_1 - \lfloor (\textbf{Entropy}_1 + 126) \div 128 \rfloor \times 2 = 111 - 2 = 109$

Determining Value

Finally, given our ‘Base’ and ‘Add’ values, we determine the final residual value as follows:

- $\textbf{is add} < 1\textbf{?}$ no
- $\textbf{p} = \textbf{log}_2(\textbf{add}) = \log_2(6) = 2$
- $\textbf{e} = 2^{p+1} - \textbf{add} - 1 = 2^3 - 6 - 1 = 1$
- $\textbf{is p} > 0\textbf{?}$ yes
- $\textbf{result} = \textbf{read 2} =$ ‘`0 0`’ $= 0$
- $\textbf{is result} > \textbf{e?}$ no
- $\textbf{sign} = \textbf{read 1} =$ ‘`0`’ $= 0$
- $\textbf{is sign} = 1\textbf{?}$ no
- $\textbf{value} = \textbf{base} + \textbf{result} = 0 + 0 = 0$

Thus, this stage consumes an additional 3 bits and our first residual value is 0.

Determining the second residual value requires going through all three steps again, with our freshly updated ‘Holding One’, ‘Holding Zero’ and ‘Entropy’ values.

Note that in a 2 channel (non-mono) block, the ‘Entropy’ values alternate between residuals. For example, Residual$_0$ uses Entropy A, Residual$_1$ uses Entropy B, Residual$_3$ uses Entropy A, and so forth. However, ‘Holding One’ and ‘Holding Zero’ are shared between channels.

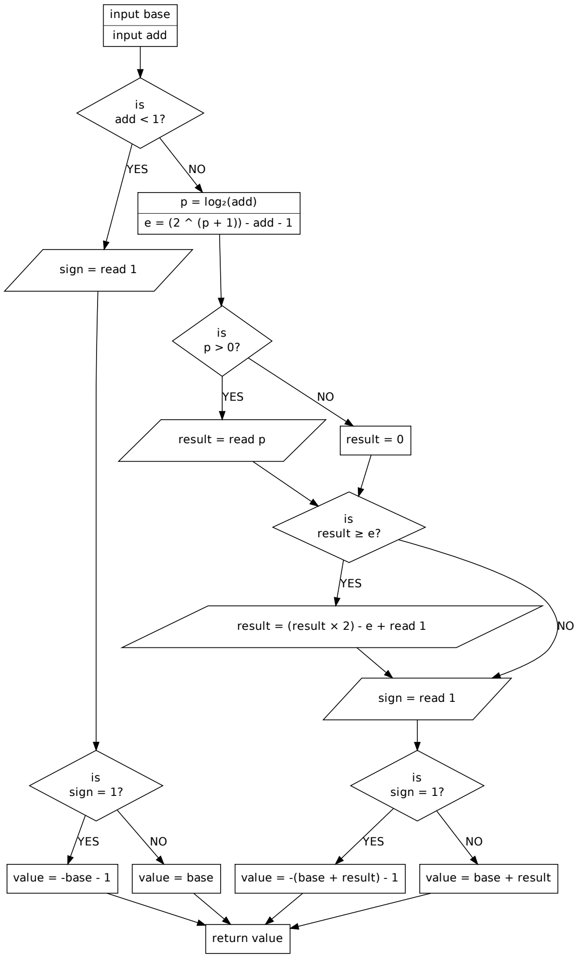


Figure 8.3: Step 3: determining value

Now, let’s run through the next residual on our remaining bits:

```
1 0 1  1 1 1 0 1 1 1 0  1 0 0 1 0 1 1 1
```

- **is holding_zero?** no
- **t = limited_unary** = ‘1 0’ = 1
- **is t = 16?** no
- **is holding_one?** yes
- **is t odd?** yes
- **holding_one = true**
- **holding_zero = false**
- $\textbf{t} = \lfloor \textbf{t} \div \textbf{2} \rfloor + \textbf{1} = \lfloor 1 \div 2 \rfloor + 1 = 1$
- **is t = 0?** no
- **is t = 1?** yes
- $\textbf{base} = \lfloor \textbf{Entropy}_1 \div 16 \rfloor + 1 = \lfloor 109 \div 16 \rfloor + 1 = 7$
- $\textbf{add} = \lfloor \textbf{Entropy}_2 \div 16 \rfloor = \lfloor 159 \div 16 \rfloor = 9$
- $\textbf{Entropy}_1 = \textbf{Entropy}_1 + \lfloor (\textbf{Entropy}_1 + 128) \div 128 \rfloor \times 5 = 114$
- $\textbf{Entropy}_2 = \textbf{Entropy}_2 - \lfloor (\textbf{Entropy}_2 + 62) \div 64 \rfloor \times 2 = 153$
- **is add** < 1**?** no
- $\textbf{p} = \textbf{log}_2(\textbf{add}) = \log_2(9) = 3$
- $\textbf{e} = 2^{p+1} - \textbf{add} - 1 = 2^4 - 9 - 1 = 6$
- **is p** > 0**?** yes
- **result = read 3** = ‘1 1 1’ = 7
- **is result** $>$ **e?** yes
- $\textbf{result} = (\textbf{result} \times 2) - \textbf{e} + \textbf{read 1} = (7 \times 2) - 6 + 1 = 9$
- **sign = read 1** = 0
- **is sign = 1?** no
- $\textbf{value} = \textbf{base} + \textbf{result} = 7 + 9 = 16$

Which returns the value 16 and consumes 7 bits in total.

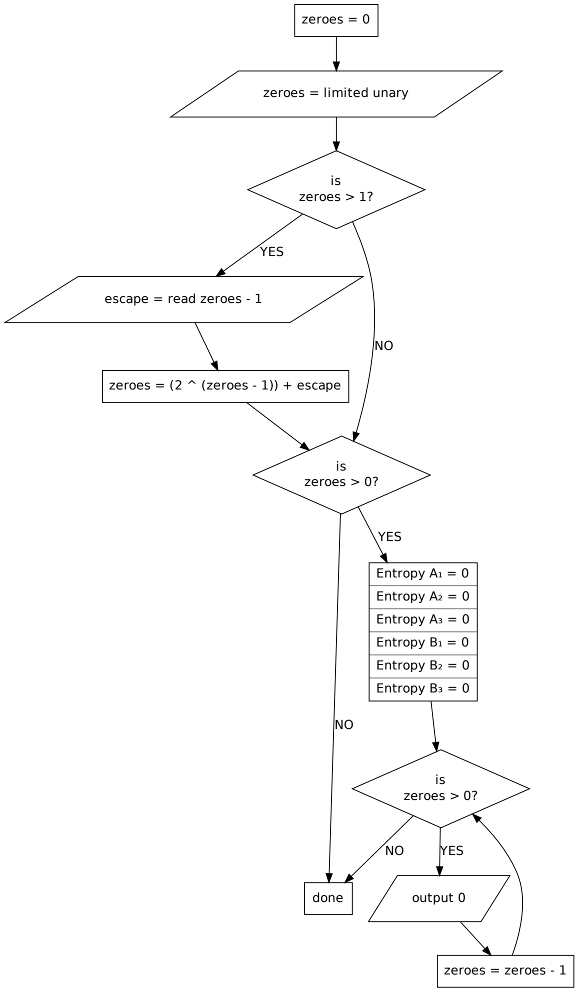
Zero Residuals

As with most other lossless codecs, WavPack features a special case to handle a large number of 0 samples in a row. This is triggered when Entropy A_1 is less than 2, Entropy B_1 is less than 2 (for non-mono blocks), and 'Holding One' and 'Holding Zero' are both false.

In that instance, we read a residual-like value to determine how many 0 values follow. If any, we set the block's six 'Entropy' variables to 0 and output the necessary number of 0 values just as regular residuals.

Therefore, for non-mono blocks, these values alternative between channels just as regular residual values do. In addition, they also count against the block's total number of samples.

Once all of the 0 values have been output, if any 'Block Samples' remain, we return to the regular residual reading process.



8.3.7 Sample Decorrelation

Once the bitstream sub-block has been decoded into a set of samples values (alternating between 'Channel A' and 'Channel B' if the block is not mono), we then apply decorrelation passes to those samples - one pass per decorrelation term value,[4] per channel.

Each decorrelation pass requires a 'Decorrelation Term', a 'Decorrelation Delta', one 'Decorrelation Weight' per channel, and one or more 'Decorrelation Sample' values - in addition to our set of input samples we're running the pass over. These passes are applied in *incrementing* order (i.e. Term_1 first, Term_2 next, and so on). The function for each pass depends on its 'Decorrelation Term':

Decorrelation Term = 18:

$$\begin{aligned}
\text{Temp}_i &= \lfloor((3 \times \text{Output}_{i-1}) - \text{Output}_{i-2}) \div 2\rfloor \\
\text{Output}_i &= \lfloor((\text{Weight}_{i-1} \times \text{Temp}_i) + 512) \div 1024\rfloor + \text{Input}_i \\
\text{Weight}_i &= \begin{cases} \text{Weight}_{i-1} & \text{if } \text{Temp}_i = 0 \text{ or } \text{Input}_i = 0 \\ \text{Weight}_{i-1} + \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Input}_i) \geq 0 \\ \text{Weight}_{i-1} - \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Input}_i) < 0 \end{cases}
\end{aligned}$$

Decorrelation Term = 17:

$$\begin{aligned}
\text{Temp}_i &= (2 \times \text{Output}_{i-1}) - \text{Output}_{i-2} \\
\text{Output}_i &= \lfloor((\text{Weight}_{i-1} \times \text{Temp}_i) + 512) \div 1024\rfloor + \text{Input}_i \\
\text{Weight}_i &= \begin{cases} \text{Weight}_{i-1} & \text{if } \text{Temp}_i = 0 \text{ or } \text{Input}_i = 0 \\ \text{Weight}_{i-1} + \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Input}_i) \geq 0 \\ \text{Weight}_{i-1} - \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Input}_i) < 0 \end{cases}
\end{aligned}$$

$1 \leq$ Decorrelation Term ≤ 8:

$$\begin{aligned}
\text{Output}_i &= \lfloor((\text{Weight}_{i-1} \times \text{Output}_{i-\text{term}}) + 512) \div 1024\rfloor + \text{Input}_i \\
\text{Weight}_i &= \begin{cases} \text{Weight}_{i-1} & \text{if } \text{Output}_{i-\text{term}} = 0 \text{ or } \text{Input}_i = 0 \\ \text{Weight}_{i-1} + \text{Delta} & \text{if } (\text{Output}_{i-\text{term}} \textbf{ xor } \text{Input}_i) \geq 0 \\ \text{Weight}_{i-1} - \text{Delta} & \text{if } (\text{Output}_{i-\text{term}} \textbf{ xor } \text{Input}_i) < 0 \end{cases}
\end{aligned}$$

Note that each function uses previously output samples for its calculation. This is where 'Decorrelation Samples' are used; those are our Output_{-1}, Output_{-2}, etc. which are used for decorrelation but not actually output.

[4] As decoded on page 74.

For 1 or 2 channel blocks, positive decorrelation terms are applied on a per-channel basis with the weight A values being applied to channel A and the weight B values being applied to channel B (if present). However, the three negative correlation terms are only valid for 2 channel blocks.

Decorrelation Term = -1:

$$\text{Output A}_i = \lfloor((\text{Weight A}_{i-1} \times \text{Output B}_{i-1}) + 512) \div 1024\rfloor + \text{Input A}_i$$

$$\text{Weight A}_i = \begin{cases} \text{Weight A}_{i-1} & \text{if Output B}_{i-1} = 0 \text{ or Input A}_i = 0 \\ \text{Weight A}_{i-1} + \text{Delta} & \text{if (Output B}_{i-1} \textbf{ xor } \text{Input A}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight A}_{i-1} - \text{Delta} & \text{if (Output B}_{i-1} \textbf{ xor } \text{Input A}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}$$

$$\text{Output B}_i = \lfloor((\text{Weight B}_{i-1} \times \text{Output A}_i) + 512) \div 1024\rfloor + \text{Input B}_i$$

$$\text{Weight B}_i = \begin{cases} \text{Weight B}_{i-1} & \text{if Output A}_i = 0 \text{ or Input B}_i = 0 \\ \text{Weight B}_{i-1} + \text{Delta} & \text{if (Output A}_i \textbf{ xor } \text{Input B}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight B}_{i-1} - \text{Delta} & \text{if (Output A}_i \textbf{ xor } \text{Input B}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}$$

Decorrelation Term = -2:

$$\text{Output B}_i = \lfloor((\text{Weight B}_{i-1} \times \text{Output A}_{i-1}) + 512) \div 1024\rfloor + \text{Input B}_i$$

$$\text{Weight B}_i = \begin{cases} \text{Weight B}_{i-1} & \text{if Output A}_{i-1} = 0 \text{ or Input B}_i = 0 \\ \text{Weight B}_{i-1} + \text{Delta} & \text{if (Output A}_{i-1} \textbf{ xor } \text{Input B}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight B}_{i-1} - \text{Delta} & \text{if (Output A}_{i-1} \textbf{ xor } \text{Input B}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}$$

$$\text{Output A}_i = \lfloor((\text{Weight A}_{i-1} \times \text{Output B}_i) + 512) \div 1024\rfloor + \text{Input A}_i$$

$$\text{Weight A}_i = \begin{cases} \text{Weight A}_{i-1} & \text{if Output B}_i = 0 \text{ or Input A}_i = 0 \\ \text{Weight A}_{i-1} + \text{Delta} & \text{if (Output B}_i \textbf{ xor } \text{Input A}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight A}_{i-1} - \text{Delta} & \text{if (Output B}_i \textbf{ xor } \text{Input A}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}$$

Decorrelation Term = -3:

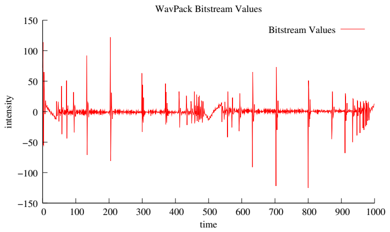
$$\text{Output A}_i = \lfloor ((\text{Weight A}_{i-1} \times \text{Output B}_{i-1}) + 512) \div 1024 \rfloor + \text{Input A}_i$$

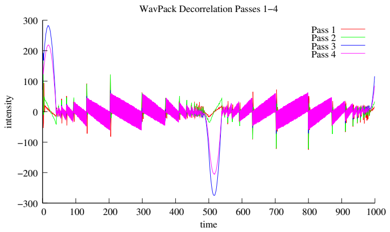
$$\text{Weight A}_i = \begin{cases} \text{Weight A}_{i-1} & \text{if Output B}_{i-1} = 0 \text{ or Input A}_i = 0 \\ \text{Weight A}_{i-1} + \text{Delta} & \text{if (Output B}_{i-1} \textbf{ xor } \text{Input A}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight A}_{i-1} - \text{Delta} & \text{if (Output B}_{i-1} \textbf{ xor } \text{Input A}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}$$

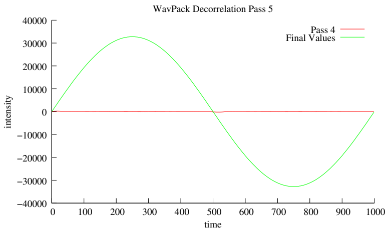
$$\text{Output B}_i = \lfloor ((\text{Weight B}_{i-1} \times \text{Output A}_{i-1}) + 512) \div 1024 \rfloor + \text{Input B}_i$$

$$\text{Weight B}_i = \begin{cases} \text{Weight B}_{i-1} & \text{if Output A}_{i-1} = 0 \text{ or Input B}_i = 0 \\ \text{Weight B}_{i-1} + \text{Delta} & \text{if (Output A}_{i-1} \textbf{ xor } \text{Input B}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight B}_{i-1} - \text{Delta} & \text{if (Output A}_{i-1} \textbf{ xor } \text{Input B}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}$$

The effect of applying these passes cumulatively is interesting when visualized on a 1 channel sine wave example stream:







Now it's time to put all this together into an example. Given a 1 channel block with the sub-block decorrelation values:

Term_1	3	Delta_1	2	Weight_1	16	$\text{Sample}_{1\ 1}$	0	$\text{Sample}_{1\ 2}$	0	$\text{Sample}_{1\ 3}$	0
Term_2	17	Delta_2	2	Weight_2	48			$\text{Sample}_{2\ 1}$	0	$\text{Sample}_{2\ 2}$	0
Term_3	2	Delta_3	2	Weight_3	32			$\text{Sample}_{3\ 1}$	0	$\text{Sample}_{3\ 2}$	0
Term_4	18	Delta_4	2	Weight_4	48			$\text{Sample}_{4\ 1}$	0	$\text{Sample}_{4\ 2}$	0
Term_5	18	Delta_5	2	Weight_5	48			$\text{Sample}_{5\ 1}$	-78	$\text{Sample}_{5\ 2}$	-73

and the residual values:

Residual_1 = -61
Residual_2 = -33

then decorrelation pass 1 applies the Term_1 formula 3, Delta_1 value of 2, $\text{Weight}_{1\ 0}$ value of 16 and initial sample values of 0 ($\text{Sample}_{1\ 1}$, $\text{Sample}_{1\ 2}$, $\text{Sample}_{1\ 3}$) to the residual input values of -61 and -33 (Channel_1 and Channel_2).

$$\begin{aligned}
\text{Output}_1 &= \lfloor((\text{Weight}_{1\ 0} \times \text{Output}_{-2}) + 512) \div 1024\rfloor + \text{Input}_1 \\
&= \lfloor((16 \times 0) + 512) \div 1024\rfloor - 61 = \textbf{-61} \\
\text{Weight}_{1\ 1} &= \text{Weight}_{1\ 0} = \textbf{16} \\
\text{Output}_2 &= \lfloor((\text{Weight}_{1\ 1} \times \text{Output}_{-1}) + 512) \div 1024\rfloor + \text{Input}_2 \\
&= \lfloor((16 \times 0) + 512) \div 1024\rfloor - 33 = \textbf{-33} \\
\text{Weight}_{1\ 2} &= \text{Weight}_{1\ 1} = \textbf{16}
\end{aligned}$$

Decorrelation pass 2 applies the Term_2 formula 17, Delta_2 value of 2, $\text{Weight}_{2\ 0}$ value of 48 and the initial sample values of 0 ($\text{Sample}_{2\ 1}$, $\text{Sample}_{2\ 2}$). Note that the inputs to pass 2 are the outputs from pass 1.

$$\begin{aligned}
\text{Temp}_1 &= (2 \times \text{Output}_0) - \text{Output}_{-1} = (2 \times 0) - 0 = \mathbf{0} \\
\text{Output}_1 &= \lfloor((\text{Weight}_{2\ 0} \times \text{Temp}_1) + 512) \div 1024\rfloor + \text{Input}_1 \\
&= \lfloor((48 \times 0) + 512) \div 1024\rfloor - 61 = \textbf{-61} \\
\text{Weight}_{2\ 1} &= \text{Weight}_{2\ 0} = \textbf{48} \\
\text{Temp}_2 &= (2 \times \text{Output}_1) - \text{Output}_0 = (2 \times -61) - 0 = \textbf{-122} \\
\text{Output}_2 &= \lfloor((\text{Weight}_{2\ 1} \times \text{Temp}_2) + 512) \div 1024\rfloor + \text{Input}_2 \\
&= \lfloor((48 \times -122) + 512) \div 1024\rfloor - 33 = -6 - 33 = \textbf{-39} \\
\text{Weight}_{2\ 2} &= \text{Weight}_{2\ 1} + \text{Delta}_2 = 48 + 2 = \textbf{50}
\end{aligned}$$

Decorrelation pass 3 applies the Term_3 formula 2, Delta_3 value of 2, $\text{Weight}_{3\ 0}$ value of 32 and initial samples of 0 ($\text{Sample}_{3\ 1}$, $\text{Sample}_{3\ 2}$).

$$\begin{aligned}
\text{Output}_1 &= \lfloor((\text{Weight}_{3\ 0} \times \text{Output}_{-1}) + 512) \div 1024\rfloor + \text{Input}_1 \\
&= \lfloor((32 \times 0) + 512) \div 1024\rfloor - 61 = \textbf{-61} \\
\text{Weight}_{3\ 1} &= \text{Weight}_{3\ 0} = \mathbf{32} \\
\text{Output}_2 &= \lfloor((\text{Weight}_{3\ 1} \times \text{Output}_0) + 512) \div 1024\rfloor + \text{Input}_2 \\
&= \lfloor((32 \times 0) + 512) \div 1024\rfloor - 39 = \textbf{-39} \\
\text{Weight}_{3\ 2} &= \text{Weight}_{3\ 1} = \mathbf{32}
\end{aligned}$$

Decorrelation pass 4 applies the Term_4 formula 18, Delta_4 value of 2, $\text{Weight}_{4\ 0}$ value of 48 and initial samples of 0 ($\text{Sample}_{4\ 1}$, $\text{Sample}_{4\ 2}$).

$$\begin{aligned}
\text{Temp}_1 &= \lfloor((3 \times \text{Output}_0) - \text{Output}_{-1}) \div 2\rfloor = \lfloor((3 \times 0) - 0) \div 2\rfloor = \mathbf{0} \\
\text{Output}_1 &= \lfloor((\text{Weight}_{4\ 0} \times \text{Temp}_1) + 512) \div 1024\rfloor + \text{Input}_1 \\
&= \lfloor((48 \times 0) + 512) \div 1024\rfloor - 61 = \textbf{-61} \\
\text{Weight}_{4\ 1} &= \text{Weight}_{4\ 0} = \mathbf{48} \\
\text{Temp}_2 &= \lfloor((3 \times \text{Output}_1) - \text{Output}_0) \div 2\rfloor = \lfloor((3 \times -61) - 0) \div 2\rfloor = \textbf{-92} \\
\text{Output}_2 &= \lfloor((\text{Weight}_{4\ 1} \times \text{Temp}_2) + 512) \div 1024\rfloor + \text{Input}_2 \\
&= \lfloor((48 \times -92) + 512) \div 1024\rfloor - 39 = -4 - 39 = \textbf{-43} \\
\text{Weight}_{4\ 2} &= \text{Weight}_{4\ 1} + \text{Delta}_4 = 48 + 2 = \mathbf{50}
\end{aligned}$$

Finally, decorrelation pass 5 applies the Term_5 formula 18, Delta_5 value of 2, $\text{Weight}_{5\ 0}$ value of 48 and initial samples of -78, -73 ($\text{Sample}_{5\ 1}$, $\text{Sample}_{5\ 2}$).

$$\begin{aligned}
\text{Temp}_1 &= \lfloor((3 \times \text{Output}_0) - \text{Output}_{-1}) \div 2\rfloor = \lfloor((3 \times -73) + 78) \div 2\rfloor = \textbf{-71} \\
\text{Output}_1 &= \lfloor((\text{Weight}_{5\ 0} \times \text{Temp}_1) + 512) \div 1024\rfloor + \text{Input}_1 \\
&= \lfloor((48 \times -71) + 512) \div 1024\rfloor - 61 = -3 - 61 = \textbf{-64} \\
\text{Weight}_{5\ 1} &= \text{Weight}_{5\ 0} + \text{Delta}_5 = 48 + 2 = \mathbf{50} \\
\text{Temp}_2 &= \lfloor((3 \times \text{Output}_1) - \text{Output}_0) \div 2\rfloor = \lfloor((3 \times -64) + 73) \div 2\rfloor = \textbf{-60} \\
\text{Output}_2 &= \lfloor((\text{Weight}_{5\ 1} \times \text{Temp}_2) + 512) \div 1024\rfloor + \text{Input}_2 \\
&= \lfloor((50 \times -60) + 512) \div 1024\rfloor - 43 = \textbf{-46} \\
\text{Weight}_{5\ 2} &= \text{Weight}_{5\ 1} + \text{Delta}_5 = 50 + 2 = \mathbf{52}
\end{aligned}$$

So, after running through all five passes, our samples are now -64 and -46.

8.3.8 Joint Stereo

If the block is not mono and the 'Joint Stereo' bit is set in the block header, our channels require one more stage of processing to transform their mid-side values back into left and right sample values.[5]

$$\begin{aligned}\text{Left}_i &= \left\lceil \frac{\text{Channel A}_i + (\text{Channel B}_i \times 2)}{2} \right\rceil \\ \text{Right}_i &= (\text{Channel B}_i \times 2) - \left\lfloor \frac{\text{Channel A}_i + (\text{Channel B}_i \times 2)}{2} \right\rfloor\end{aligned}$$

For example, given the Channel A samples of -64 and -46, and the Channel B samples of 32 and 39, we convert them to left and right samples as follows:

$$\begin{aligned}\text{Left}_1 &= \left\lceil \frac{\text{Channel A}_1 + (\text{Channel B}_1 \times 2)}{2} \right\rceil = \left\lceil \frac{-64 + (32 \times 2)}{2} \right\rceil = \mathbf{0} \\ \text{Right}_1 &= (\text{Channel B}_1 \times 2) - \left\lfloor \frac{\text{Channel A}_1 + (\text{Channel B}_1 \times 2)}{2} \right\rfloor \\ &= (32 \times 2) - \left\lfloor \frac{-64 + (32 \times 2)}{2} \right\rfloor = 64 - 0 = \mathbf{64} \\ \text{Left}_2 &= \left\lceil \frac{-46 + (39 \times 2)}{2} \right\rceil = \mathbf{16} \\ \text{Right}_2 &= (\text{Channel B}_2 \times 2) - \left\lfloor \frac{\text{Channel A}_2 + (\text{Channel B}_2 \times 2)}{2} \right\rfloor \\ &= (39 \times 2) - \left\lfloor \frac{-46 + (39 \times 2)}{2} \right\rfloor = 78 - 16 = \mathbf{62}\end{aligned}$$

Thus, our left samples are 0 and 16, and our right samples are 64 and 62.

8.3.9 the CRC

Verifying the block's CRC is quite simple:

$$\text{CRC}_i = (3 \times \text{CRC}_{i-1}) + \text{Decoded Sample}_i$$

where Decoded Sample_i alternates between channels if necessary, CRC_{-1} begins with a value of `0xFFFFFFFF` and each CRC_i value is truncated to 32 bits.

The CRC is calculated *after* the joint stereo transformation, but *before* handling extended/shifted integers and false stereo.

[5] In the case of multi-channel audio, these aren't necessarily *front* left and right; they might be side left and right or rear left and right channels.

8.3.10 Extended/Shifted Integers

If ‘Extended Size Integers’ is set in the block header, there should be an ‘Int32 Info’ sub-block present whose layout is as follows:

Metadata Function (9)	0	Actual Size 1 Less	Large Block (0)	Block Size (2)
0 4	5	6	7	8 15

Sent Bits	Zero Bits	One Bits	Duplicate Bits
16 23	24 31	32 39	40 47

Curiously, these values are exclusive; if ‘Zero Bits’ is present, ‘One Bits’ is ignored and so forth. If ‘Zero Bits’ is non-zero, we pad each sample’s least-significant bits with that many 0 bits. If ‘One Bits’ is non-zero, we pad each sample’s least-significant bits with that many 1 bits. If ‘Duplicate Bits’ is non-zero, we pad each sample’s least-significant bits with that sample’s own least-significant bit, ‘Duplicate Bits’ number of times.

This can be summarized as follows:

$$\text{Extended}_i = \begin{cases} \text{Original}_i \times 2^{\text{Zero Bits}} & \text{if `Zero Bits'} \ > 0 \\ \text{Original}_i \times 2^{\text{One Bits}} + (2^{\text{One Bits}} - 1) & \text{if `One Bits'} \ > 0 \\ \text{Original}_i \times 2^{\text{Duplicate Bits}} & \\ \quad \text{if `Duplicate Bits'} \ > 0 \text{ and } \text{Original}_i \bmod 2 = 0 & \\ \text{Original}_i \times 2^{\text{Duplicate Bits}} + (2^{\text{Duplicate Bits}} - 1) & \\ \quad \text{if `Duplicate Bits'} \ > 0 \text{ and } \text{Original}_i \bmod 2 = 1 & \end{cases}$$

8.3.11 Channel Info

A WavPack file with more than 2 channels should have a ‘Channel Info’ sub-block to indicate its channel layout.

Metadata Function (0xD)	0	Actual Size 1 Less	Large Block (0)	Block Size
0 4	5	6	7	8 15

Channel Count	Channel Mask
16 23	24

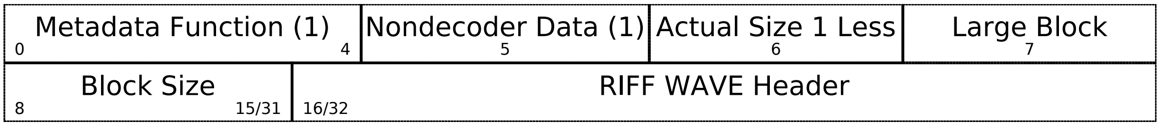
‘Channel Mask’ is the same as used by Wave, as shown on page 21.

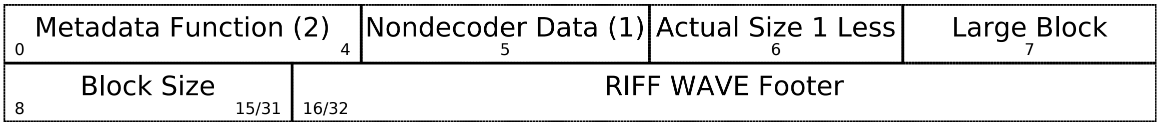
8.3.12 False Stereo

If the ‘False Stereo’ bit is set in the block header, we’ve been treating the block as being mono thus far. At this point, we duplicate Channel A’s values to Channel B just prior to returning the from the block.

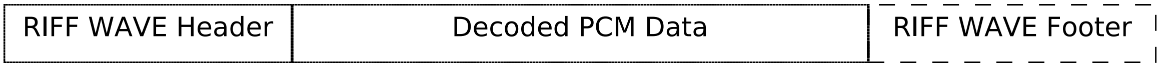
8.3.13 RIFF WAVE Header/Footer

These sub-blocks are typically found in the first and last WavPack block, respectively. The header must always be present in the file while the footer is optional.



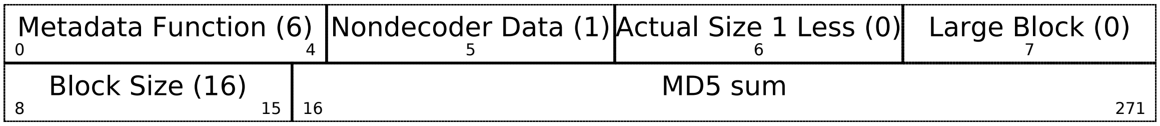


One can think of them as halves of a ‘PCM sandwich’ of which our decoded data comprises the ‘meat’:



8.3.14 MD5

This optional sub-block is typically found in the final WavPack block.



The MD5 is the hash of all the samples over the entire file. It is calculated by running the hashing algorithm[6] over the raw input samples in little-endian format and signed if their bits-per-sample are greater than 8.

[6] As described by RFC1321

8.4 WavPack Encoding

For WavPack encoding, one needs a stream of input PCM values along with the stream's sample rate, number of channels, bits per sample and channel mask.

We first split our input samples into chunks containing 'Block Size' number of PCM frames. Since WavPack's headers are relatively large and its adaptive algorithm is quite good over long stretches of samples, it makes sense to use a large block size. The reference encoder defaults to 44100 PCM frames.

The next step is to split those chunks of PCM frames into WavPack blocks containing 1 or 2 channels each. For a one channel input stream, the blocks are sent as follows:

Block	First Block Bit	Last Block Bit	Is Mono	Channel A	Channel B
Block_1	1	1	1	Front Center	

For a two channel input stream, the blocks are sent as follows:

Block	First Block Bit	Last Block Bit	Is Mono	Channel A	Channel B
Block_1	1	1	0	Front Left	Front Right

However, for multi-channel input streams, we need to split its channels into a set of blocks with 1 or 2 channels per block. By using the channel mask[7] we can split the stream into 2 channel blocks with left-right channel pairs and 1 channel blocks for everything else.

For example, given a 6-channel audio stream with the channel mask `0x3F`, we have the channels 'Front Left', 'Front Right', 'Front Center', 'LFE', 'Back Left' and 'Back Right' - in that order. So, a good way to split our channels into blocks is as follows:

Block	First Block Bit	Last Block Bit	Is Mono	Channel A	Channel B
Block_1	1	0	0	Front Left	Front Right
Block_2	0	0	1	Front Center	
Block_3	0	0	1	LFE	
Block_4	0	1	0	Back Left	Back Right

8.4.1 Channel Info

If the stream has more than 2 channels, a 'Channel Info' sub-block should be added to the first block, as illustrated on page 91.

8.4.2 False Stereo

If the block is stereo and Channel A's samples are identical to Channel B's samples, one can set the 'False Stereo' bit in the block header and treat the block as having only one

[7] As explained on page 21.

channel for the rest of its encoding. Note that the block's 'Is Mono' bit is still `false` in this case.

8.4.3 Extended/Shifted Integers

If the following condition holds:

$$0 = \sum_{i=0}^{\text{block size}-1} \text{Channel}_i \bmod 2^{bits}$$

for Channel A and, if present, Channel B where $bits > 0$, then the highest value of $bits$ if what's used for the 'Zero Bits' field in an 'Extended Size Integers' sub-block, as described on page 91. Each channel's samples are then divided by 2^{bits} for the remainder of encoding and the 'Extended Size Integers' bit is set in the block header.

8.4.4 the CRC

After the audio samples have been processed for false stereo and wasted bits, it's best to perform the block header CRC calculation before starting to encode them, as follows:

$$\text{CRC}_i = (3 \times \text{CRC}_{i-1}) + \text{Sample}_i$$

where Sample_i alternates between channels if necessary. CRC_{-1} begins with a value of `0xFFFFFFFF` and each CRC_i value is truncated to 32 bits.

8.4.5 Joint Stereo

Next, for two channel blocks, one typically converts both channels to joint stereo. This involves transforming independent left and right channels to mid and side channels.

$$\begin{aligned} \text{Mid}_i &= \text{Channel A}_i - \text{Channel B}_i \\ \text{Side}_i &= \left\lfloor \frac{\text{Channel A}_i + \text{Channel B}_i}{2} \right\rfloor \end{aligned}$$

Where 'Mid' is the new 'Channel A' and 'Side' is the new 'Channel B'. For example, given the 'Channel A' value of 16 and the 'Channel B' value of 62, our conversion is as follows:

$$\begin{aligned} \text{Mid}_0 &= 16 - 62 = \textbf{-46} \\ \text{Side}_0 &= \left\lfloor \frac{16 + 62}{2} \right\rfloor = \textbf{39} \end{aligned}$$

One must also set the 'Joint Stereo' bit in the block header.

8.4.6 Block Header

Once the ‘False Stereo’, ‘Extended Size Integers’, ‘Joint Stereo’ and ‘CRC’ values are decided, we can finally write a block header based on our input:

Block ID `wvpk' (0x6B707677) 0 31	Block Size 32 63		
Version 64 79	Track Number 80 87	Index Number 88 95	Total Samples 96 127
Block Index 128 159	Block Samples 160 191		
Bits per Sample 192 193	Mono Output 194	Hybrid Mode 195	Joint Stereo 196
Channel Decorrelation 197	Hbd. Noise Shaping 198	Floating Point Data 199	Extended Size Integers 200
Hbd. Controls Bitrate 201	Hbd. Noise Balanced 202	Initial Block 203	Final Block 204
Left Shift Data 205 209	Maximum Magnitude 210 214	Sample Rate 215 218	
Reserved 219 220	Use IIR 221	False Stereo 222	Reserved 223
CRC 224 255			

The remaining fields are as follows:

Block Size 24 + byte length of sub blocks

Version 0x407

Track Number 0

Index Number 0

Block Index total PCM frames written thus far

Block Samples total PCM frames of block

Hybrid Mode 0

Channel Decorrelation 1 if stereo, 0 if mono

Hybrid Noise Shaping 0

Floating Point Data 0

Hybrid Controls Bitrate 0

Hybrid Noise Balanced 0

Left Shift Data 0

sample rate	value
6000	0000
8000	0001
9600	0010
11025	0011
12000	0100
16000	0101
22050	0110
24000	0111
32000	1000
44100	1001
48000	1010
64000	1011
88200	1100
96000	1101
192000	1110

bits per sample	value
8	00
16	01
24	10
32	11

Maximum Magnitude maximum sample size, in bits

Use IIR 0

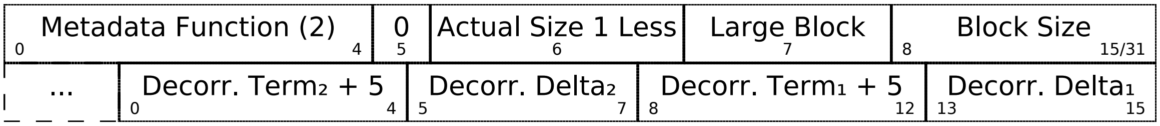
Note that the ‘Block Size’ and ‘Total Samples’ fields can’t be known in advance; all the block’s sub-blocks must be generated before we’ll know the former, and the entire file must be written before we’ll know the latter.

8.4.7 Decorrelation Terms/Deltas

These are typically defined by the number of decorrelation passes to use:

Terms	Decorrelation Passes					Deltas	Decorrelation Passes				
	1	2	5	10	16		1	2	5	10	16
Term_1	18	17	3	4	2	Delta_1	2	2	2	2	2
Term_2		18	17	17	18	Delta_2		2	2	2	2
Term_3			2	-1	-1	Delta_3			2	2	2
Term_4			18	5	8	Delta_4			2	2	2
Term_5			18	3	6	Delta_5			2	2	2
Term_6				2	3	Delta_6				2	2
Term_7				-2	5	Delta_7				2	2
Term_8				18	7	Delta_8				2	2
Term_9				18	4	Delta_9				2	2
Term_{10}				18	2	Delta_{10}				2	2
Term_{11}					18	Delta_{11}					2
Term_{12}					-2	Delta_{12}					2
Term_{13}					3	Delta_{13}					2
Term_{14}					2	Delta_{14}					2
Term_{15}					18	Delta_{15}					2
Term_{16}					18	Delta_{16}					2

They are placed in a sub-block as follows:



Metadata Function (2)	0	Actual Size 1 Less	Large Block	Block Size
0 4	5	6	7	8 15/31

...	Decorr. Term_2 + 5	Decorr. Delta_2	Decorr. Term_1 + 5	Decorr. Delta_1
	0 4	5 7	8 12	13 15

Since each term/delta pair is 8 bits, ‘Actual Size 1 Less’ is set when the number of terms is odd, ‘Large Block’ is always going to be 0 and ‘Block Size’ equals the number of terms, divided by 2.

8.4.8 Decorrelation Passes

Once our number of decorrelation passes is decided, we must also generate decorrelation weights, decorrelation samples and entropy variables sub-blocks before moving on to the residuals sub-block. So where do we get those values? They actually come from the *previous* block.[8] Since encoding will modify decorrelation weights and entropy variables as it progresses, the final values for Block_i become the initial values for Block_{i+1}. As for the decorrelation values, the final few decorrelated samples (whose quantity depends on the decorrelation term) are 'wrapped' from the previous decorrelation pass into our 'Decorrelation Samples' sub-block as its starting point.

However, we can't store the previous block's final values as-is. Remember that the values for decorrelation weights are multiplied by 2^3 and the values for decorrelation samples and entropy variables are stored logarithmically. Therefore, we must 'round-trip' the previous block's output samples before using them as input samples since they'll be parsed the same way during decoding. This process will be explained in the sub-block sections to follow.

For Block_0, we'll set our initial decorrelation weights, decorrelation samples and entropy variables to 0.

The application of each pass requires a 'Decorrelation Term', a 'Decorrelation Delta', one 'Decorrelation Weight' per channel and one or more 'Decorrelation Sample' values - in addition to the set of processed input samples we're running the pass over.

Decorrelation Term = 18:

$$\begin{aligned}
\text{Temp}_i &= \lfloor((3 \times \text{Input}_{i-1}) - \text{Input}_{i-2}) \div 2\rfloor \\
\text{Output}_i &= \text{Input}_i - \lfloor((\text{Weight}_{i-1} \times \text{Temp}_i) + 512) \div 1024\rfloor \\
\text{Weight}_i &= \begin{cases} \text{Weight}_{i-1} & \text{if } \text{Temp}_i = 0 \text{ or } \text{Output}_i = 0 \\ \text{Weight}_{i-1} + \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Output}_i) \geq 0 \\ \text{Weight}_{i-1} - \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Output}_i) < 0 \end{cases}
\end{aligned}$$

Decorrelation Term = 17:

$$\begin{aligned}
\text{Temp}_i &= (2 \times \text{Input}_{i-1}) - \text{Input}_{i-2} \\
\text{Output}_i &= \text{Input}_i - \lfloor((\text{Weight}_{i-1} \times \text{Temp}_i) + 512) \div 1024\rfloor \\
\text{Weight}_i &= \begin{cases} \text{Weight}_{i-1} & \text{if } \text{Temp}_i = 0 \text{ or } \text{Output}_i = 0 \\ \text{Weight}_{i-1} + \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Output}_i) \geq 0 \\ \text{Weight}_{i-1} - \text{Delta} & \text{if } (\text{Temp}_i \textbf{ xor } \text{Output}_i) < 0 \end{cases}
\end{aligned}$$

[8] More precisely, the previous block covering the same set of channels - in the case of multi-channel audio.

$1 \leq$ Decorrelation Term ≤ 8:

$$\text{Output}_i = \text{Input}_i - \lfloor((\text{Weight}_{i-1} \times \text{Input}_{i-\text{term}}) + 512) \div 1024\rfloor$$

$$\text{Weight}_i = \begin{cases} \text{Weight}_{i-1} & \text{if Input}_{i-\text{term}} = 0 \text{ or Output}_i = 0 \\ \text{Weight}_{i-1} + \text{Delta} & \text{if (Input}_{i-\text{term}} \textbf{ xor } \text{Output}_i) \geq 0 \\ \text{Weight}_{i-1} - \text{Delta} & \text{if (Input}_{i-\text{term}} \textbf{ xor } \text{Output}_i) < 0 \end{cases}$$

Similar to decoding, each function uses previous input samples for its calculation. This is where ‘Decorrelation Samples’ are used; those are our Input_{-1}, Input_{-2}, etc. which are used for decorrelation but not actually output.

For 1 or 2 channel blocks, positive decorrelation terms are applied on a per-channel basis with the weight A values being applied to channel A and the weight B values being applied to channel B (if present). However, the three negative correlation terms are only valid for 2 channel blocks:

Decorrelation Term = -1:

$$\begin{aligned}
\text{Temp A}_i &= \text{Input B}_{i-1} \\
\text{Temp B}_i &= \text{Input A}_i \\
\text{Output A}_i &= \text{Input A}_i - \lfloor((\text{Weight A}_{i-1} \times \text{Temp A}_i) + 512) \div 1024\rfloor \\
\text{Weight A}_i &= \begin{cases} \text{Weight A}_{i-1} & \text{if Temp A}_i \text{ or Output A}_i = 0 \\ \text{Weight A}_{i-1} + \text{Delta} & \text{if (Temp A}_i \textbf{ xor } \text{Output A}_i) \geq 0 \\ \ \text{to a maximum of 1024} & \\ \text{Weight A}_{i-1} - \text{Delta} & \text{if (Temp A}_i \textbf{ xor } \text{Output A}_i) < 0 \\ \ \text{to a minimum of -1024} & \end{cases} \\
\text{Output B}_i &= \text{Input B}_i - \lfloor((\text{Weight B}_{i-1} \times \text{Temp B}_i) + 512) \div 1024\rfloor \\
\text{Weight B}_i &= \begin{cases} \text{Weight B}_{i-1} & \text{if Temp B}_i \text{ or Output B}_i = 0 \\ \text{Weight B}_{i-1} + \text{Delta} & \text{if (Temp B}_i \textbf{ xor } \text{Output B}_i) \geq 0 \\ \ \text{to a maximum of 1024} & \\ \text{Weight B}_{i-1} - \text{Delta} & \text{if (Temp B}_i \textbf{ xor } \text{Output B}_i) < 0 \\ \ \text{to a minimum of -1024} & \end{cases}
\end{aligned}$$

Decorrelation Term = -2:

$$
\begin{aligned}
\text{Temp A}_i &= \text{Input B}_i \\
\text{Temp B}_i &= \text{Input A}_{i-1} \\
\text{Output A}_i &= \text{Input A}_i - \lfloor ((\text{Weight A}_{i-1} \times \text{Temp A}_i) + 512) \div 1024 \rfloor \\
\text{Weight A}_i &= \begin{cases} \text{Weight A}_{i-1} & \text{if Temp A}_i \text{ or Output A}_i = 0 \\ \text{Weight A}_{i-1} + \text{Delta} & \text{if (Temp A}_i \textbf{ xor } \text{Output A}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight A}_{i-1} - \text{Delta} & \text{if (Temp A}_i \textbf{ xor } \text{Output A}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases} \\
\text{Output B}_i &= \text{Input B}_i - \lfloor ((\text{Weight B}_{i-1} \times \text{Temp B}_i) + 512) \div 1024 \rfloor \\
\text{Weight B}_i &= \begin{cases} \text{Weight B}_{i-1} & \text{if Temp B}_i \text{ or Output B}_i = 0 \\ \text{Weight B}_{i-1} + \text{Delta} & \text{if (Temp B}_i \textbf{ xor } \text{Output B}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight B}_{i-1} - \text{Delta} & \text{if (Temp B}_i \textbf{ xor } \text{Output B}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}
\end{aligned}
$$

Decorrelation Term = -3:

$$
\begin{aligned}
\text{Temp A}_i &= \text{Input B}_{i-1} \\
\text{Temp B}_i &= \text{Input A}_{i-1} \\
\text{Output A}_i &= \text{Input A}_i - \lfloor ((\text{Weight A}_{i-1} \times \text{Temp A}_i) + 512) \div 1024 \rfloor \\
\text{Weight A}_i &= \begin{cases} \text{Weight A}_{i-1} & \text{if Temp A}_i \text{ or Output A}_i = 0 \\ \text{Weight A}_{i-1} + \text{Delta} & \text{if (Temp A}_i \textbf{ xor } \text{Output A}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight A}_{i-1} - \text{Delta} & \text{if (Temp A}_i \textbf{ xor } \text{Output A}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases} \\
\text{Output B}_i &= \text{Input B}_i - \lfloor ((\text{Weight B}_{i-1} \times \text{Temp B}_i) + 512) \div 1024 \rfloor \\
\text{Weight B}_i &= \begin{cases} \text{Weight B}_{i-1} & \text{if Temp B}_i \text{ or Output B}_i = 0 \\ \text{Weight B}_{i-1} + \text{Delta} & \text{if (Temp B}_i \textbf{ xor } \text{Output B}_i) \geq 0 \\ \text{ to a maximum of 1024} & \\ \text{Weight B}_{i-1} - \text{Delta} & \text{if (Temp B}_i \textbf{ xor } \text{Output B}_i) < 0 \\ \text{ to a minimum of -1024} & \end{cases}
\end{aligned}
$$

8.4.9 Decorrelation Weights

Once the decorrelation passes for Block$_0$ have been completed (with its initial decorrelation weight values of 0), we should store its final updated weight values to be used as the initial decorrelation weights for Block$_1$, as so on through the rest of the file.

There is one decorrelation weight value per decorrelation pass, per channel. Each has a minimum value of -1024 and a maximum value of 1024. Converting their values to 8 bits requires the following formula:

$$\text{value} = \begin{cases} \left\lfloor (\text{Weight} - \left\lfloor \frac{\text{Weight}+2^6}{7} \right\rfloor + 4) \div 2^3 \right\rfloor & \text{if Weight} > 0 \\ 0 & \text{if Weight} = 0 \\ \left\lfloor (\text{Weight} + 4) \div 2^3 \right\rfloor & \text{if Weight} < 0 \end{cases}$$

Weights are placed in a sub-block in reverse order as follows:

Metadata Function (3) 0 4	0 5	Actual Size 1 Less 6	Large Block 7	Block Size 8 15/31

...	Decorrelation Weight$_2$ 0 7	Decorrelation Weight$_1$ 8 15

Since each decorrelation weight value is stored in 8 bits, 'Actual Size 1 Less' is set if the total number of weights is odd, 'Large Block' is always going to be 0 and 'Block Size' is the total number of weights divided by 2.

After the initial weights for Block$_i$ have been stored, the 'round-trip' formula to retrieve those weight values for Block$_i$'s decorrelation passes is as follows:

$$\text{Decorrelation Weight} = \begin{cases} \text{value} \times 2^3 + \left\lfloor \frac{\text{value} \times 2^3 + 2^6}{2^7} \right\rfloor & \text{if value} > 0 \\ 0 & \text{if value} = 0 \\ \text{value} \times 2^3 & \text{if value} < 0 \end{cases}$$

8.4.10 Decorrelation Samples

We apply the following formulas to convert our 32-bit, signed decorrelation values to 16-bit signed sub-block values:

$$\begin{aligned}
asample &= |sample| + \left\lfloor \frac{|sample|}{2^9} \right\rfloor \\
bitcount &= \text{count\_bits}(asample) \\
\text{Value} &= \begin{cases}
(bitcount \times 2^8) + \text{wv\_log2}((asample \times 2^{9-bitcount}) \bmod 256) \\
\text{ if } 0 \leq asample < 256 \text{ and } sample \geq 0 \\
(bitcount \times 2^8) + \text{wv\_log2}(\lfloor asample \div 2^{bitcount-9} \rfloor \bmod 256) \\
\text{ if } 256 \leq asample \text{ and } sample \geq 0 \\
-((bitcount \times 2^8) + \text{wv\_log2}((asample \times 2^{9-bitcount}) \bmod 256)) \\
\text{ if } 0 \leq asample < 256 \text{ and } sample < 0 \\
-((bitcount \times 2^8) + \text{wv\_log2}(\lfloor asample \div 2^{bitcount-9} \rfloor \bmod 256)) \\
\text{ if } 256 \leq asample \text{ and } sample < 0
\end{cases}
\end{aligned}$$

where ‘count_bits’ is defined as follows:

$$\text{count\_bits(x)} = \begin{cases} 0 & \text{if } x = 0 \\ 1 + \text{count\_bits}(\lfloor x \div 2 \rfloor) & \text{if } x \neq 0 \end{cases}$$

and ‘wv_log2’ is defined from the following base-16 table:

	0x?0	0x?1	0x?2	0x?3	0x?4	0x?5	0x?6	0x?7	0x?8	0x?9	0x?A	0x?B	0x?C	0x?D	0x?E	0x?F
0x0?	0x00	0x01	0x03	0x04	0x06	0x07	0x09	0x0a	0x0b	0x0d	0x0e	0x10	0x11	0x12	0x14	0x15
0x1?	0x16	0x18	0x19	0x1a	0x1c	0x1d	0x1e	0x20	0x21	0x22	0x24	0x25	0x26	0x28	0x29	0x2a
0x2?	0x2c	0x2d	0x2e	0x2f	0x31	0x32	0x33	0x34	0x36	0x37	0x38	0x39	0x3b	0x3c	0x3d	0x3e
0x3?	0x3f	0x41	0x42	0x43	0x44	0x45	0x47	0x48	0x49	0x4a	0x4b	0x4d	0x4e	0x4f	0x50	0x51
0x4?	0x52	0x54	0x55	0x56	0x57	0x58	0x59	0x5a	0x5c	0x5d	0x5e	0x5f	0x60	0x61	0x62	0x63
0x5?	0x64	0x66	0x67	0x68	0x69	0x6a	0x6b	0x6c	0x6d	0x6e	0x6f	0x70	0x71	0x72	0x74	0x75
0x6?	0x76	0x77	0x78	0x79	0x7a	0x7b	0x7c	0x7d	0x7e	0x7f	0x80	0x81	0x82	0x83	0x84	0x85
0x7?	0x86	0x87	0x88	0x89	0x8a	0x8b	0x8c	0x8d	0x8e	0x8f	0x90	0x91	0x92	0x93	0x94	0x95
0x8?	0x96	0x97	0x98	0x99	0x9a	0x9b	0x9b	0x9c	0x9d	0x9e	0x9f	0xa0	0xa1	0xa2	0xa3	0xa4
0x9?	0xa5	0xa6	0xa7	0xa8	0xa9	0xa9	0xaa	0xab	0xac	0xad	0xae	0xaf	0xb0	0xb1	0xb2	0xb2
0xA?	0xb3	0xb4	0xb5	0xb6	0xb7	0xb8	0xb9	0xb9	0xba	0xbb	0xbc	0xbd	0xbe	0xbf	0xc0	0xc0
0xB?	0xc1	0xc2	0xc3	0xc4	0xc5	0xc6	0xc6	0xc7	0xc8	0xc9	0xca	0xcb	0xcb	0xcc	0xcd	0xce
0xC?	0xcf	0xd0	0xd0	0xd1	0xd2	0xd3	0xd4	0xd4	0xd5	0xd6	0xd7	0xd8	0xd8	0xd9	0xda	0xdb
0xD?	0xdc	0xdc	0xdd	0xde	0xdf	0xe0	0xe0	0xe1	0xe2	0xe3	0xe4	0xe4	0xe5	0xe6	0xe7	0xe7
0xE?	0xe8	0xe9	0xea	0xea	0xeb	0xec	0xed	0xee	0xee	0xef	0xf0	0xf1	0xf1	0xf2	0xf3	0xf4
0xF?	0xf4	0xf5	0xf6	0xf7	0xf7	0xf8	0xf9	0xf9	0xfa	0xfb	0xfc	0xfc	0xfd	0xfe	0xff	0xff

For example, given a sample value of 28:

$$\begin{aligned}
asample &= |28| + \left\lfloor \frac{|28|}{2^9} \right\rfloor = 28 + 0 = \mathbf{28} \\
bitcount &= \mathbf{5} \\
value &= (5 \times 2^8) + \text{wv\_log2}((28 \times 2^{9-5}) \bmod 256) \\
&= 1280 + \text{wv\_log2}(448 \bmod 256) \\
&= 1280 + \text{wv\_log2}(192) = \mathbf{1487}
\end{aligned}$$

These samples are then placed in a sub-block as follows:

Metadata Function (4)	0	Actual Size 1 Less	Large Block	Block Size
0 4	5	6	7	8 15/31

Decorrelation Sample$_1$	Decorrelation Sample$_2$	...
0 15	16 31	

where ‘Actual Size 1 Less’ and ‘Large Block’ are 0, and ‘Block Size’ is the total number of decorrelation samples.

Writing the values themselves requires traversing the decorrelation samples lists in *reverse* order, from i = ‘Decorrelation Passes’ $- 1$ to 0.

For Stereo Block

- If $17 \leq$ Decorrelation Term$_i$ ≤ 18
  1. Write Sample $A_{i\ 1}$
  2. Write Sample $A_{i\ 0}$
  3. Write Sample $B_{i\ 1}$
  4. Write Sample $B_{i\ 0}$
- If $1 \leq$ Decorrelation Term$_i$ ≤ 8
  1. For $j = 0$ to Decorrelation Term$_i$ $- 1$
     a) Write Sample $A_{i\ j}$
     b) Write Sample $B_{i\ j}$
- If $-3 \leq$ Decorrelation Term$_i$ ≤ -1
  1. Write Sample $B_{i\ 0}$
  2. Write Sample $A_{i\ 0}$

For Mono Block

- If $17 \leq$ Decorrelation Term$_i$ ≤ 18
  1. Write Sample $A_{i\ 1}$
  2. Write Sample $A_{i\ 0}$
- If $1 \leq$ Decorrelation Term$_i$ ≤ 8
  1. For $j = 0$ to Decorrelation Term$_i$ $- 1$
     a) Write Sample $A_{i\ j}$

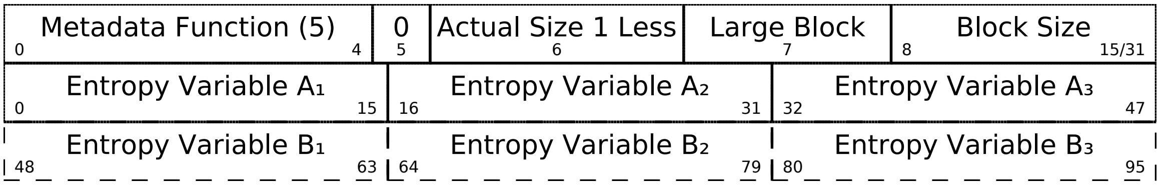
Round-tripping these values back to decorrelation samples for the next block requires applying the same formula as decoding:

$$
\text{Sample} = \begin{cases}
\lfloor \text{wv\_exp2}(\mathit{value} \bmod 256) \div 2^{9-\lfloor \mathit{value} \div 2^8 \rfloor} \rfloor & \text{if } 0 \leq \mathit{value} \leq 2304 \\
\text{wv\_exp2}(\mathit{value} \bmod 256) \times 2^{\lfloor \mathit{value} \div 2^8 \rfloor - 9} & \text{if } 2304 < \mathit{value} \leq 32767 \\
-\lfloor \text{wv\_exp2}(-\mathit{value} \bmod 256) \div 2^{9-\lfloor -\mathit{value} \div 2^8 \rfloor} \rfloor & \text{if } -2304 \leq \mathit{value} < 0 \\
-(\text{wv\_exp2}(-\mathit{value} \bmod 256) \times 2^{\lfloor -\mathit{value} \div 2^8 \rfloor - 9}) & \text{if } -32768 \leq \mathit{value} < -2304
\end{cases}
$$

where 'wv_exp2' is defined from the following base-16 table:

	0x?0	0x?1	0x?2	0x?3	0x?4	0x?5	0x?6	0x?7	0x?8	0x?9	0x?A	0x?B	0x?C	0x?D	0x?E	0x?F
0x0?	100	101	101	102	103	103	104	105	106	106	107	108	108	109	10A	10B
0x1?	10B	10C	10D	10E	10E	10F	110	110	111	112	113	113	114	115	116	116
0x2?	117	118	119	119	11A	11B	11C	11D	11D	11E	11F	120	120	121	122	123
0x3?	124	124	125	126	127	128	128	129	12A	12B	12C	12C	12D	12E	12F	130
0x4?	130	131	132	133	134	135	135	136	137	138	139	13A	13A	13B	13C	13D
0x5?	13E	13F	140	141	141	142	143	144	145	146	147	148	148	149	14A	14B
0x6?	14C	14D	14E	14F	150	151	151	152	153	154	155	156	157	158	159	15A
0x7?	15B	15C	15D	15E	15E	15F	160	161	162	163	164	165	166	167	168	169
0x8?	16A	16B	16C	16D	16E	16F	170	171	172	173	174	175	176	177	178	179
0x9?	17A	17B	17C	17D	17E	17F	180	181	182	183	184	185	187	188	189	18A
0xA?	18B	18C	18D	18E	18F	190	191	192	193	195	196	197	198	199	19A	19B
0xB?	19C	19D	19F	1A0	1A1	1A2	1A3	1A4	1A5	1A6	1A8	1A9	1AA	1AB	1AC	1AD
0xC?	1AF	1B0	1B1	1B2	1B3	1B4	1B6	1B7	1B8	1B9	1BA	1BC	1BD	1BE	1BF	1C0
0xD?	1C2	1C3	1C4	1C5	1C6	1C8	1C9	1CA	1CB	1CD	1CE	1CF	1D0	1D2	1D3	1D4
0xE?	1D6	1D7	1D8	1D9	1DB	1DC	1DD	1DE	1E0	1E1	1E2	1E4	1E5	1E6	1E8	1E9
0xF?	1EA	1EC	1ED	1EE	1F0	1F1	1F2	1F4	1F5	1F6	1F8	1F9	1FA	1FC	1FD	1FF

8.4.11 the Entropy Variables Sub-Block



'Actual Size 1 Less' and 'Large Block' are 0. 'Block Size' is 3 for mono blocks and 6 for stereo blocks. The samples themselves are converted and round-tripped the same way that 'Decorrelation Sample' values are, as explained on pages 101 and 76.

8.4.12 the Bitstream Sub-Block

Given a set of residual values and one set of 3 entropy variables per channel, the final encoding step for a WavPack block is generating the bitstream sub-block. Since the subframe header requires a size, we must either write it in advance with a size of 0 and rewrite it once the sub-block is finished, or first write the residuals to temporary space before writing the sub-block header.

Metadata Function (0xA) 0 4	0 5	Actual Size 1 Less 6	Large Block 7	Block Size 8 15/31
Residual Data 32				

As with decoding, writing each residual value is a multi-stage process which involves calculating a unary value, referencing and updating the channel's set of entropy variables, and calculating a fixed collection of bits.

However, this procedure is complicated by the 'holding_one' and 'holding_zero' boolean values. As you'll recall, when 'holding_zero' is false, the decoder skips reading a unary value entirely. This means that before we can output Residual_{i-1}'s values, we must determine the values for Residual_i so that the 'holding_one' and 'holding_zero' boolean values can be set properly. Note that holding_one_{-1} and holding_zero_{-1} are both `false`.

In practice, we'll first need to handle the special case of many zero residuals in a row as discussed on page 108. But for clarity, it's best to understand the general case first.

Calculate Unsigned Value and Sign Bit

$$\text{value}_i = \begin{cases} \text{Residual}_i & \text{if } \text{Residual}_i \geq 0 \\ -\text{Residual}_i - 1 & \text{if } \text{Residual}_i < 0 \end{cases}$$

$$\text{sign}_i = \begin{cases} 0 & \text{if } \text{Residual}_i \geq 0 \\ 1 & \text{if } \text{Residual}_i < 0 \end{cases}$$

Calculate Unary Value, High, Low and Next Medians

$$\begin{aligned} \text{Median}_{i\ 0} &= \lfloor \text{Entropy}_{i\ 0} \div 2^4 \rfloor + 1 \\ \text{Median}_{i\ 1} &= \lfloor \text{Entropy}_{i\ 1} \div 2^4 \rfloor + 1 \\ \text{Median}_{i\ 2} &= \lfloor \text{Entropy}_{i\ 2} \div 2^4 \rfloor + 1 \end{aligned}$$

If $\text{value}_i < \text{Median}_{i\ 0}$:

$$\begin{aligned}
\text{unary}_i &= 0\\
\text{low}_i &= 0\\
\text{high}_i &= \text{Median}_{i\ 0} - 1\\
\text{Entropy}_{(i+1)\ 0} &= \text{Entropy}_{i\ 0} - \left\lfloor \frac{\text{Entropy}_{i\ 0} + 126}{128} \right\rfloor \times 2\\
\text{Entropy}_{(i+1)\ 1} &= \text{Entropy}_{i\ 1}\\
\text{Entropy}_{(i+1)\ 2} &= \text{Entropy}_{i\ 2}
\end{aligned}$$

If $(\text{value}_i - \text{Median}_{i\ 0}) < \text{Median}_{i\ 1}$:

$$\begin{aligned}
\text{unary}_i &= 1\\
\text{low}_i &= \text{Median}_{i\ 0}\\
\text{high}_i &= \text{Median}_{i\ 0} + \text{Median}_{i\ 1} - 1\\
\text{Entropy}_{(i+1)\ 0} &= \text{Entropy}_{i\ 0} + \left\lfloor \frac{\text{Entropy}_{i\ 0} + 128}{128} \right\rfloor \times 5\\
\text{Entropy}_{(i+1)\ 1} &= \text{Entropy}_{i\ 1} - \left\lfloor \frac{\text{Entropy}_{i\ 1} + 62}{64} \right\rfloor \times 2\\
\text{Entropy}_{(i+1)\ 2} &= \text{Entropy}_{i\ 2}
\end{aligned}$$

If $(\text{value}_i - (\text{Median}_{i\ 0} + \text{Median}_{i\ 1})) < \text{Median}_{i\ 2}$:

$$\begin{aligned}
\text{unary}_i &= 2\\
\text{low}_i &= \text{Median}_{i\ 0} + \text{Median}_{i\ 1}\\
\text{high}_i &= \text{Median}_{i\ 0} + \text{Median}_{i\ 1} + \text{Median}_{i\ 2} - 1\\
\text{Entropy}_{(i+1)\ 0} &= \text{Entropy}_{i\ 0} + \left\lfloor \frac{\text{Entropy}_{i\ 0} + 128}{128} \right\rfloor \times 5\\
\text{Entropy}_{(i+1)\ 1} &= \text{Entropy}_{i\ 1} + \left\lfloor \frac{\text{Entropy}_{i\ 1} + 64}{64} \right\rfloor \times 5\\
\text{Entropy}_{(i+1)\ 2} &= \text{Entropy}_{i\ 2} - \left\lfloor \frac{\text{Entropy}_{i\ 2} + 30}{32} \right\rfloor \times 2
\end{aligned}$$

Otherwise:

$$\begin{aligned}
\text{unary}_i &= \left\lfloor \frac{\text{value}_i - (\text{Median}_{i\ 0} + \text{Median}_{i\ 1})}{\text{Median}_{i\ 2}} \right\rfloor + 2 \\
\text{low}_i &= \text{Median}_{i\ 0} + \text{Median}_{i\ 1} + ((\text{unary}_i - 2) \times \text{Median}_{i\ 2}) \\
\text{high}_i &= \text{low}_i + \text{Median}_{i\ 2} - 1 \\
\text{Entropy}_{(i+1)\ 0} &= \text{Entropy}_{i\ 0} + \left\lfloor \frac{\text{Entropy}_{i\ 0} + 128}{128} \right\rfloor \times 5 \\
\text{Entropy}_{(i+1)\ 1} &= \text{Entropy}_{i\ 1} + \left\lfloor \frac{\text{Entropy}_{i\ 1} + 64}{64} \right\rfloor \times 5 \\
\text{Entropy}_{(i+1)\ 2} &= \text{Entropy}_{i\ 2} + \left\lfloor \frac{\text{Entropy}_{i\ 2} + 32}{32} \right\rfloor \times 5
\end{aligned}$$

Calculate Fixed Size, Fixed Value and Optional Extra Bit

If $\text{low}_i = \text{high}_i$:

$$\begin{aligned}
\text{fixed size}_i &= 0 \\
\text{has extra}_i &= \texttt{false}
\end{aligned}$$

Otherwise:

$$\text{extras}_i = 2^{\text{count\_bits}(\text{high}_i - \text{low}_i)} - (\text{high}_i - \text{low}_i) - 1$$

If $\text{low}_i \neq \text{high}_i$ and $(\text{value}_i - \text{low}_i) < \text{extras}_i$:

$$\begin{aligned}
\text{fixed size}_i &= \text{value}_i - \text{low}_i \\
\text{fixed value}_i &= \text{count\_bits}(\text{high}_i - \text{low}_i) - 1 \\
\text{has extra}_i &= \texttt{false}
\end{aligned}$$

If $\text{low}_i \neq \text{high}_i$ and $(\text{value}_i - \text{low}_i) \geq \text{extras}_i$:

$$\begin{aligned}
\text{fixed size}_i &= \lfloor (\text{value}_i - \text{low}_i + \text{extras}_i) \div 2 \rfloor \\
\text{fixed value}_i &= \text{count\_bits}(\text{high}_i - \text{low}_i) - 1 \\
\text{has extra}_i &= \texttt{true} \\
\text{extra bit}_i &= (\text{value}_i - \text{low}_i + \text{extras}_i) \bmod 2
\end{aligned}$$

Update Previous Residual Based On Current Residual

If $\text{unary}_{i-1} > 0$ and $\text{unary}_i > 0$:

$$\text{unary}_{i-1} \leftarrow \begin{cases} (\text{unary}_{i-1} \times 2) + 1 & \text{if holding one}_{i-1} = \texttt{false} \\ (\text{unary}_{i-1} \times 2) - 1 & \text{if holding one}_{i-1} = \texttt{true} \end{cases}$$
$$\text{holding zero}_i = \texttt{false}$$
$$\text{holding one}_i = \texttt{true}$$

If $\text{unary}_{i-1} = 0$ and $\text{unary}_i > 0$:

$$\text{unary}_{i-1} \leftarrow \begin{cases} 1 & \text{if holding zero}_{i-1} = \texttt{false} \\ \textit{not output} & \text{if holding zero}_{i-1} = \texttt{true} \end{cases}$$
$$\text{holding zero}_i = \texttt{false}$$
$$\text{holding one}_i = \texttt{not}\ \text{holding zero}_{i-1}$$

If $\text{unary}_{i-1} > 0$ and $\text{unary}_i = 0$:

$$\text{unary}_{i-1} \leftarrow \begin{cases} \text{unary}_{i-1} \times 2 & \text{if holding one}_{i-1} = \texttt{false} \\ (\text{unary}_{i-1} - 1) \times 2 & \text{if holding one}_{i-1} = \texttt{true} \end{cases}$$
$$\text{holding zero}_i = \texttt{true}$$
$$\text{holding one}_i = \texttt{false}$$

If $\text{unary}_{i-1} = 0$ and $\text{unary}_i = 0$:

$$\text{unary}_{i-1} \leftarrow \begin{cases} 0 & \text{if holding zero}_{i-1} = \texttt{false} \\ \textit{not output} & \text{if holding zero}_{i-1} = \texttt{true} \end{cases}$$
$$\text{holding zero}_i = \texttt{not}\ \text{holding zero}_{i-1}$$
$$\text{holding one}_i = \texttt{false}$$

Output Previous Residual

Once the previous residual's unary value has been determined, its fields can now be output.

- If unary output, write unary_{i-1} number of `1` bits followed by a `0` bit (if $\text{unary}_{i-1} < 16$)
- Write 'fixed size_{i-1}' number of bits with the value 'fixed value_{i-1}'
- If 'has extra_{i-1}', write a single bit with the value 'extra bit_{i-1}'

- Write a single bit with the value ‘sign_{i-1}’

Note that if $\text{unary}_{i-1} \geq 16$, we write an escape code instead.

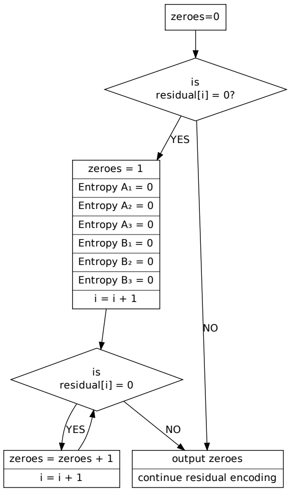
- If $\text{unary}_{i-1} = 16$, write 18 bits with the value `0xFFFF` (unary 16 plus unary 0)
- If $\text{unary}_{i-1} = 17$, write 19 bits with the value `0x2FFFF` (unary 16 plus unary 1)
- If $\text{unary}_{i-1} \geq 18$, write 17 bits with the value `0xFFFF` (unary 16)
  - Write $\text{count\_bits}(\text{unary}_{i-1} - 16)$ number of `1` bits followed by a `0` bit
  - Write $\text{count\_bits}(\text{unary}_{i-1} - 16) - 1$ number of bits with the value $((\text{unary}_{i-1} - 16) \bmod 2^{\text{count\_bits}(\text{unary}_{i-1}-16)-1})$

Handle Groups Of Zero Residuals

This is necessary when Entropy $A_{i\ 0} < 2$ and, for 2 channel blocks, Entropy $B_{i\ 0} < 2$, holding_zero_i = `false` and holding_one_i = `false`. In that event, whether the current Residual_i is 0 or not, we must generate a zeroes block after outputting Residual_{i-1} but before outputting Residual_i.

Once the number of zeroes has been determined, their output is quite similar to an escaped unary value.

- If zeroes = 0, write a single `0` bit.
- If zeroes > 0
  - Write count_bits(zeroes) number of `1` bits followed by a `0` bit
  - Write $\text{count\_bits}(\text{zeroes}) - 1$ number of bits with the value $(\text{zeroes} \bmod 2^{\text{count\_bits}(\text{zeroes})-1})$



8.4.13 Extended Integers

In the rare case that a block has ‘wasted bits’, as explained on page 94, we generate the following sub-block to store our ‘Zero Bits’ value:

Metadata Function (9) 0 4	0 5	Actual Size 1 Less 6	Large Block (0) 7	Block Size (2) 8 15

Sent Bits 16 23	Zero Bits 24 31	One Bits 32 39	Duplicate Bits 40 47

8.4.14 RIFF WAVE Header

WavPack expects to find a RIFF WAVE header sub-block in the first block within the file. This sub-block is laid out as follows:

<table>
<tr><td>Metadata Function (1)
0 4</td><td>Nondecoder Data (1)
5</td><td>Actual Size 1 Less
6</td><td>Large Block
7</td></tr>
<tr><td>Block Size
8 15/31</td><td colspan="3">RIFF WAVE Header
16/32</td></tr>
</table>

The RIFF WAVE header is everything from the start of a RIFF WAVE file to the end of its `data` chunk's header. For non WAVEFORMATEXTENSIBLE files, this is typically the first 36 bytes. For WAVEFORMATEXTENSIBLE files, this is typically the first 60 bytes.

8.4.15 the Footer Block

Though not required, WavPack files often contain a trailing block after the audio has been exhausted. This block contains only an MD5 sum sub-block and optional RIFF WAVE footer sub-block for wave files with additional chunks of data after the `data` chunk.

RIFF WAVE Footer

<table>
<tr><td>Metadata Function (2)
0 4</td><td>Nondecoder Data (1)
5</td><td>Actual Size 1 Less
6</td><td>Large Block
7</td></tr>
<tr><td>Block Size
8 15/31</td><td colspan="3">RIFF WAVE Footer
16/32</td></tr>
</table>

MD5 Sum

<table>
<tr><td>Metadata Function (6)
0 4</td><td>Nondecoder Data (1)
5</td><td>Actual Size 1 Less (0)
6</td><td>Large Block (0)
7</td></tr>
<tr><td>Block Size (16)
8 15</td><td colspan="3">MD5 sum
16 271</td></tr>
</table>

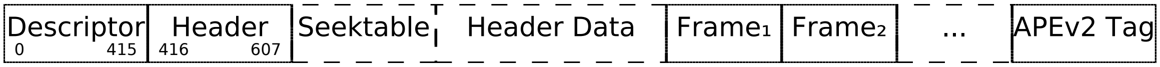
The MD5 is the hash of all the samples over the entire file. It is calculated by running the hashing algorithm[9] over the raw input samples in little-endian format and signed if their bits-per-sample are greater than 8.

[9] As described by RFC1321

9 Monkey's Audio

Monkey's Audio is a lossless RIFF WAVE compressor. Unlike FLAC, which is a PCM compressor, Monkey's Audio also stores IFF chunks and reproduces the original WAVE file in its entirety rather than storing only the data it contains. All of its fields are little-endian.

9.1 the Monkey's Audio File Stream



9.2 the Monkey's Audio Descriptor

ID (`MAC' 0x4D414320) 0 31	Version 32 63	
Descriptor Bytes 64 95	Header Bytes 96 127	Seektable Bytes 128 159
Header Data Bytes 160 191	Frame Data Bytes 192 223	Frame Data Bytes (High) 224 255
Terminating Data Bytes 256 287	MD5 Sum 288 415	

'Version' is the encoding software's version times 1000. i.e. Monkey's Audio 3.99 = 3990

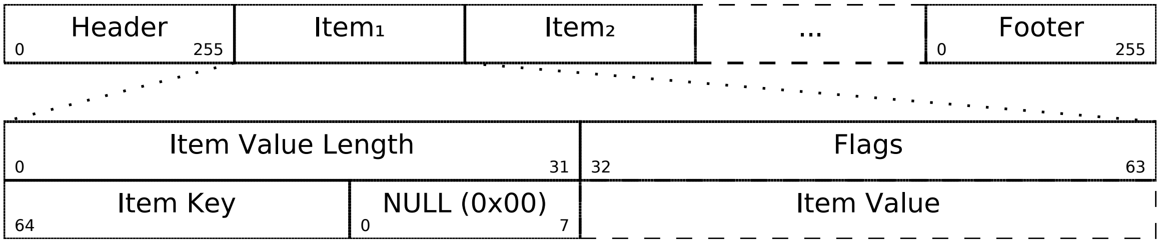
9.3 the Monkey's Audio header

Compression Level 0 15	Format Flags 16 31
Blocks Per Frame 32 63	Final Frame Blocks 64 95
Total Frames 96 127	Bits Per Sample 128 143
Channels 144 159	Sample Rate 160 191

$$\text{Length in Seconds} = \frac{((\text{Total Frames} - 1) \times \text{Blocks Per Frame}) + \text{Final Frame Blocks}}{\text{Sample Rate}} \tag{9.1}$$

9.4 the APEv2 Tag

The APEv2 tag is a little-endian metadata tag appended to Monkey's Audio files, among others.



'Item Key' is an ASCII string from the range 0x20 to 0x7E. 'Item Value' is typically a UTF-8 encoded string, but may also be binary depending on the Flags.

Abstract	Abstract
Album	album name
Artist	performing artist
Bibliography	Bibliography/Discography
Catalog	catalog number
Comment	user comment
Composer	original composer
Conductor	conductor
Copyright	copyright holder
Debut album	debut album name
Dummy	place holder
EAN/UPC	EAN-13/UPC-A bar code identifier
File	file location
Genre	genre
Index	indexes for quick access
Introplay	characteristic part of piece for intro playing
ISBN	ISBN number with check digit
ISRC	International Standard Recording Number
Language	used Language(s) for music/spoken words
LC	Label Code
Media	source media
Publicationright	publication right holder
Publisher	record label or publisher
Record Date	record date
Record Location	record location
Related	location of related information
Subtitle	track subtitle
Title	track title
Track	track number
Year	release date

9.4.1 the APEv2 Tag Header/Footer

<table>
<tr><td colspan="6">Preamble (`APETAGEX' 0x4150455441474558)
0 63</td></tr>
<tr><td colspan="3">Version (0xD0070000)
64 95</td><td colspan="3">Tag Size
96 127</td></tr>
<tr><td colspan="2">Item Count
128 159</td><td colspan="2">Flags
160 191</td><td colspan="2">Reserved
192 255</td></tr>
</table>

The format of the APEv2 header and footer are identical except for the 'Is Header' tag. 'Version' is typically 2000 (stored little-endian). 'Tag Size' is the size of the entire APEv2 tag, including the footer but excluding the header. 'Item Count' is the number of individual tag items.

9.4.2 the APEv2 Flags

<table>
<tr><td colspan="2">Undefined (0x00)
0 4</td><td>Encoding
5 6</td><td>Read-Only
7</td></tr>
<tr><td colspan="4">Undefined (0x00)
8 23</td></tr>
<tr><td>Container Header
24</td><td>Contains no Footer
25</td><td>Is Header
26</td><td>Undefined (0x00)
27 31</td></tr>
</table>

This flags field is used by both the APEv2 header/footer and the individual tag items. The 'Encoding' field indicates the encoding of its value:

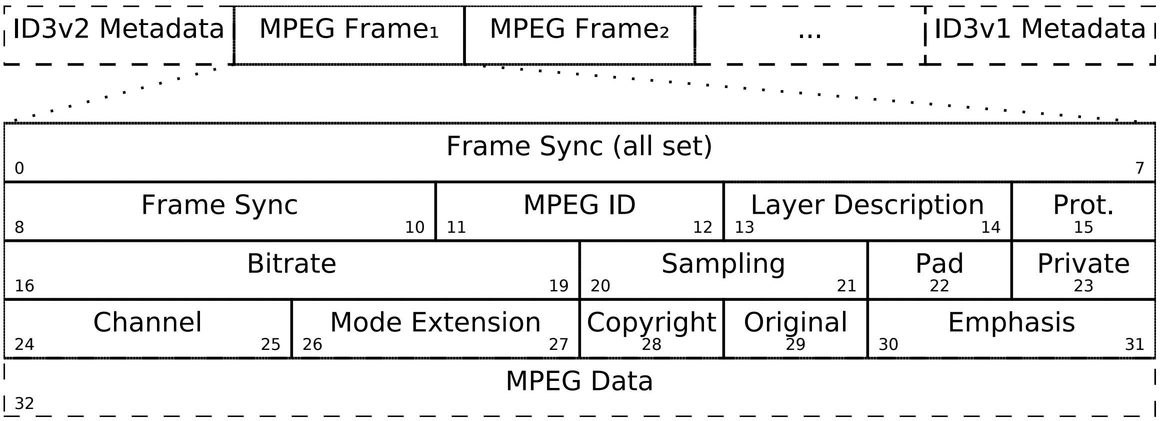
`00` = UTF-8, `01` = Binary, `10` = External Link, `11` = Reserved .

10 MP3

MP3 is the de-facto standard for lossy audio. It is little more than a series of MPEG frames with an optional ID3v2 metadata header and optional ID3v1 metadata footer.

MP3 decoders are assumed to be very tolerant of anything in the stream that doesn't look like an MPEG frame, ignoring such junk until the next frame is found. Since MP3 files have no standard container format in which non-MPEG data can be placed, metadata such as ID3 tags are often made 'sync-safe' by formatting them in a way that decoders won't confuse tags for MPEG frames.

10.1 the MP3 File Stream



			Sample Rate			
bits	MPEG ID	Description	MPEG-1	MPEG-2	MPEG-2.5	Channels
00	MPEG-2.5	reserved	44100	22050	11025	Stereo
01	reserved	Layer III	48000	24000	12000	Joint stereo
10	MPEG-2	Layer II	32000	16000	8000	Dual channel stereo
11	MPEG-1	Layer I	reserved	reserved	reserved	Mono

Layer I frames always contain 384 samples. Layer II and Layer III frames always contain 1152 samples. If the 'Protection' bit is 0, the frame header is followed by a 16 bit CRC.

bits	MPEG-1 Layer-1	MPEG-1 Layer-2	MPEG-1 Layer-3	MPEG-2 Layer-1	MPEG-2 Layer-2/3
0000	free	free	free	free	free
0001	32	32	32	32	8
0010	64	48	40	48	16
0011	96	56	48	56	24
0100	128	64	56	64	32
0101	160	80	64	80	40
0110	192	96	80	96	48
0111	224	112	96	112	56
1000	256	128	112	128	64
1001	288	160	128	144	80
1010	320	192	160	160	96
1011	352	224	192	176	112
1100	384	256	224	192	128
1101	416	320	256	224	144
1110	448	384	320	256	160
1111	bad	bad	bad	bad	bad

Table 10.1: Bitrate in 1000 bits per second

To find the total size of an MPEG frame, use one of the following formulas:

Layer I:

$$\text{Byte Length} = \left(\frac{12 \times \text{Bitrate}}{\text{Sample Rate}} + \text{Pad}\right) \times 4 \tag{10.1}$$

Layer II/III:

$$\text{Byte Length} = \frac{144 \times \text{Bitrate}}{\text{Sample Rate}} + \text{Pad} \tag{10.2}$$

For example, an MPEG-1 Layer III frame with a sampling rate of 44100, a bitrate of 128kbps and a set pad bit is 418 bytes long, including the header.

$$\frac{144 \times 128000}{44100} + 1 = 418 \tag{10.3}$$

10.1.1 the Xing Header

An MP3 frame header contains the track's sampling rate, bits-per-sample and number of channels. However, because MP3 files are little more than concatenated MPEG frames, there is no obvious place to store the track's total length. Since the length of each frame is a constant number of samples, one can calculate the track length by counting the number of frames. This method is the most accurate but is also quite slow.

For MP3 files in which all frames have the same bitrate - also known as constant bitrate, or CBR files - one can divide the total size of file (minus any ID3 headers/footers), by the bitrate to determine its length. If an MP3 file has no Xing header in its first frame, one can assume it is CBR.

An MP3 file that does contain a Xing header in its first frame can be assumed to be variable bitrate, or VBR. In that case, the rate of the first frame cannot be used as a basis to calculate the length of the entire file. Instead, one must use the information from the Xing header which contains that length.

All of the fields within a Xing header are big-endian.

Header `Xing' (0x58696E67)	0	31
Flags	32	63
Number of Frames	64	95
Bytes	96	127
TOC Entry$_1$	128	135
TOC Entry$_2$	136	143
...	144	919
TOC Entry$_{100}$	920	927
Quality	928	959

10.2 ID3v1 Tags

ID3v1 tags are very simple metadata tags appended to an MP3 file. All of the fields are fixed length and the text encoding is undefined. There are two versions of ID3v1 tags. ID3v1.1 has a track number field as a 1 byte value at the end of the comment field. If the byte just before the end is not null (0x00), assume we're dealing with a classic ID3v1 tag without a track number.

10.2.1 ID3v1

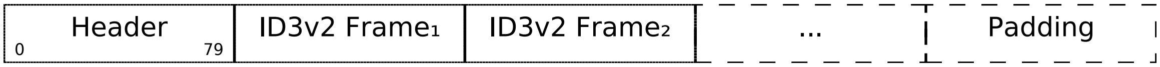
Header (`TAG' 0x544147)	0	23
Track Title	24	263
Artist Name	264	503
Album Name	504	743
Year	744	775
Comment	776	1015
Genre	1016	1023

10.2.2 ID3v1.1

Header (`TAG' 0x544147)	0	23
Track Title	24	263
Artist Name	264	503
Album Name	504	743
Year	744	775
Comment	776	999
NULL	1000	1007
Track Number	1008	1015
Genre	1016	1023

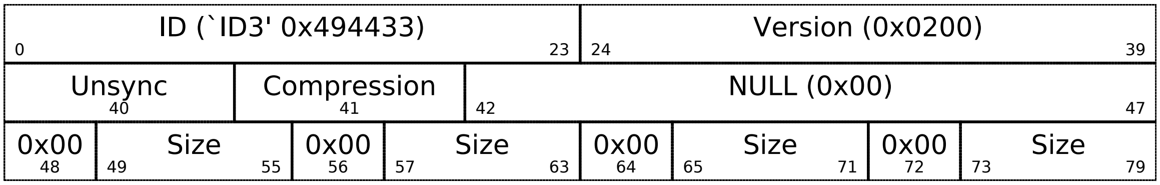
10.3 ID3v2 Tags

The ID3v2 tag was invented to address the deficiencies in the original ID3v1 tag. ID3v2 comes in three similar but not entirely compatible variants: ID3v2.2, ID3v2.3 and ID3v2.4. All of its fields are big-endian.



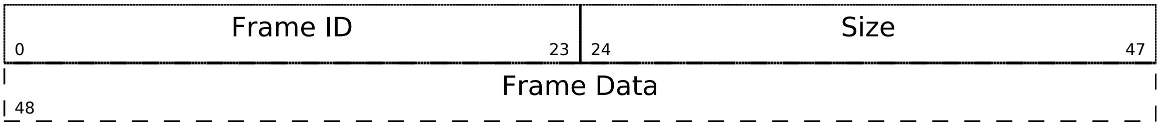
10.3.1 ID3v2.2

ID3v2.2 Header

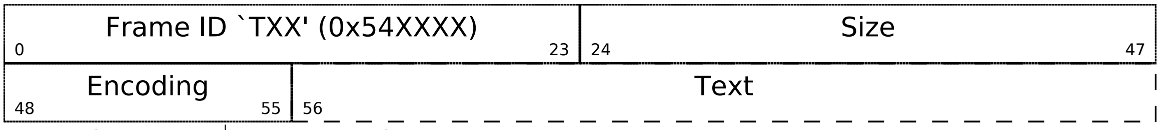


The single Size field is split by NULL (0x00) bytes in order to make it 'sync-safe'. That is, no possible size value will result in a false MP3 frame sync (11 bits set in a row). This size field is the length of the entire tag, not including the header.

ID3v2.2 Frame



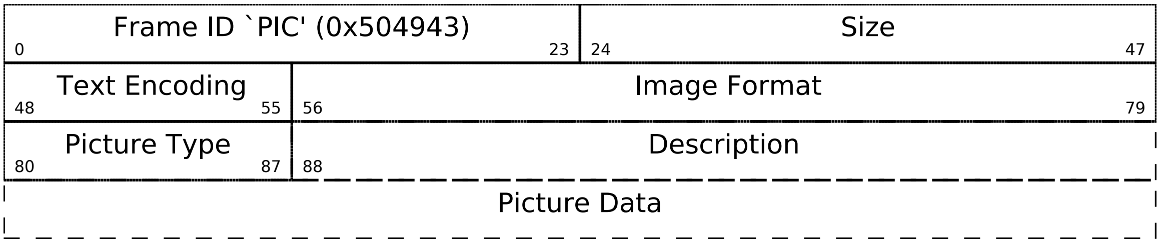
Frame IDs that begin with the letter 'T' (0x54) are text frames. These have an additional text encoding byte before the actual text data. All text strings may be terminated by a null character (0x00 or 0x0000, depending on the encoding).



Encoding Byte	Text Encoding
0x00	ISO-8859-1
0x01	UCS-16

ID3v2.2 PIC Frame

‘PIC’ frames are attached pictures. This allows an ID3v2.2 tag to contain a JPEG or PNG image, typically of album artwork which can be displayed to the user when the track is played.



Text Encoding is the encoding of the Description field. Its value is either ISO-8859-1 or UCS-16 - the same as in text frames. Image Format is a 3 byte string indicating the format of the image, typically ‘JPG’ for JPEG images or ’PNG’ for PNG images. Description is a NULL-terminated C-string which contains a text description of the image.

value	type	value	type
0	Other	1	32x32 pixels ‘file icon’ (PNG only)
2	Other file icon	3	Cover (front)
4	Cover (back)	5	Leaflet page
6	Media (e.g. label side of CD)	7	Lead artist / Lead performer / Soloist
8	Artist / Performer	9	Conductor
10	Band / Orchestra	11	Composer
12	Lyricist / Text writer	13	Recording location
14	During recording	15	During performance
16	Movie / Video screen capture	17	A bright colored fish
18	Illustration	19	Band / Artist logotype
20	Publisher / Studio logotype		

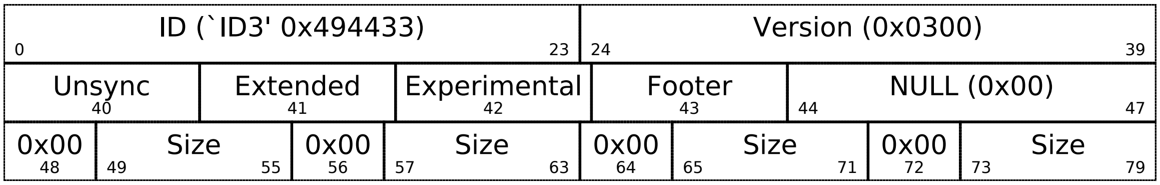
Table 10.2: PIC image types

ID3v2.2 Frame IDs

BUF	Recommended buffer size
CNT	Play counter
COM	Comments
CRA	Audio encryption
CRM	Encrypted meta frame
ETC	Event timing codes
EQU	Equalization
GEO	General encapsulated object
IPL	Involved people list
LNK	Linked information
MCI	Music CD Identifier
MLL	MPEG location lookup table
PIC	Attached picture
POP	Popularimeter
REV	Reverb
RVA	Relative volume adjustment
SLT	Synchronized lyric/text
STC	Synced tempo codes
TAL	Album/Movie/Show title
TBP	BPM (Beats Per Minute)
TCM	Composer
TCO	Content type
TCR	Copyright message
TDA	Date
TDY	Playlist delay
TEN	Encoded by
TFT	File type
TIM	Time
TKE	Initial key
TLA	Language(s)
TLE	Length
TMT	Media type
TOA	Original artist(s) / performer(s)
TOF	Original filename
TOL	Original Lyricist(s) / text writer(s)
TOR	Original release year
TOT	Original album / Movie / Show title
TP1	Lead artist(s) / performer(s) / Soloist(s) / Performing group
TP2	Band / Orchestra / Accompaniment
TP3	Conductor / Performer refinement
TP4	Interpreted, remixed, or otherwise modified by
TPA	Part of a set
TPB	Publisher
TRC	ISRC (International Standard Recording Code)
TRD	Recording dates
TRK	Track number / Position in set
TSI	Size
TSS	Software / hardware and settings used for encoding
TT1	Content group description
TT2	Title / Songname / Content description
TT3	Subtitle / Description refinement
TXT	Lyricist / text writer
TXX	User defined text information frame
TYE	Year
UFI	Unique file identifier
ULT	Unsychronized lyric / text transcription
WAF	Official audio file webpage
WAR	Official artist / performer webpage
WAS	Official audio source webpage
WCM	Commercial information
WCP	Copyright / Legal information
WPB	Publishers official webpage
WXX	User defined URL link frame

10.3.2 ID3v2.3

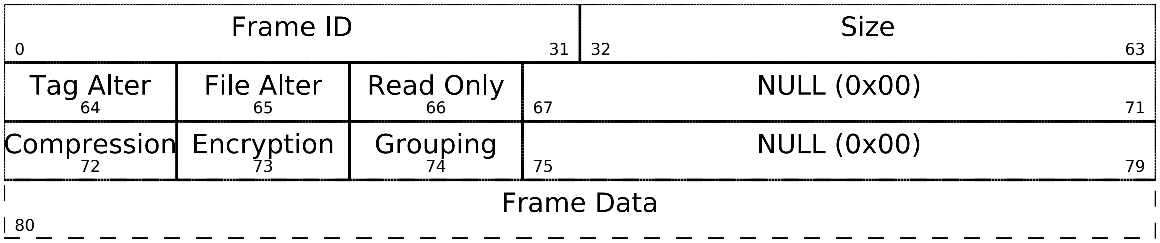
ID3v2.3 Header



ID (`ID3' 0x494433) 0 23				Version (0x0300) 24 39			
Unsync 40	Extended 41	Experimental 42	Footer 43	NULL (0x00) 44 47			
0x00 48	Size 49 55	0x00 56	Size 57 63	0x00 64	Size 65 71	0x00 72	Size 73 79

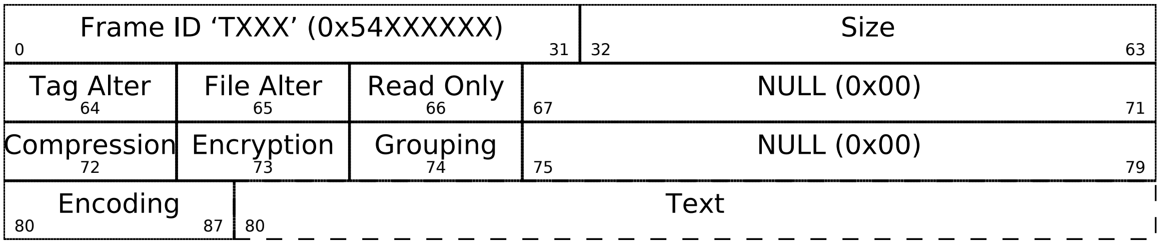
The single Size field is split by NULL (0x00) bytes in order to make it 'sync-safe'. This size field is the length of the entire tag, not including the header.

ID3v2.3 Frame



Frame ID 0 31		Size 32 63	
Tag Alter 64	File Alter 65	Read Only 66	NULL (0x00) 67 71
Compression 72	Encryption 73	Grouping 74	NULL (0x00) 75 79
Frame Data 80			

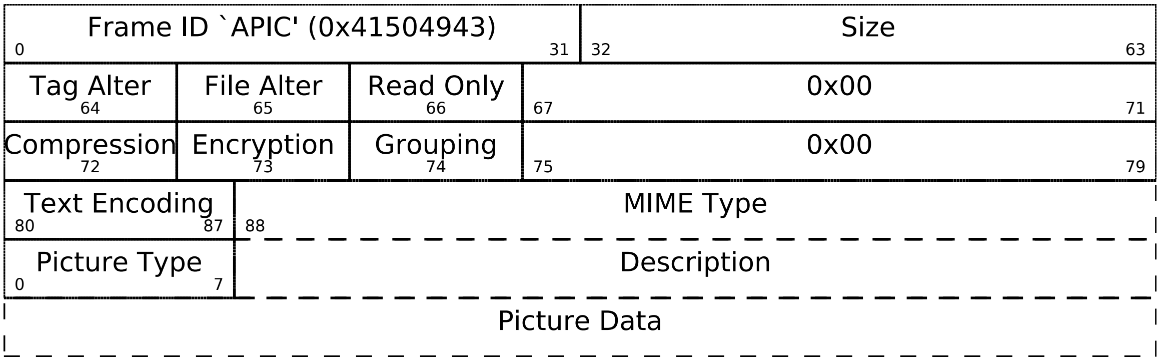
Frame IDs that begin with the letter 'T' (0x54) are text frames. These have an additional text encoding byte before the actual text data. All text strings may be terminated by a null character (0x00 or 0x0000, depending on the encoding).



Frame ID 'TXXX' (0x54XXXXXX) 0 31		Size 32 63	
Tag Alter 64	File Alter 65	Read Only 66	NULL (0x00) 67 71
Compression 72	Encryption 73	Grouping 74	NULL (0x00) 75 79
Encoding 80 87	Text 80		

Encoding Byte	Text Encoding
0x00	ISO-8859-1
0x01	UCS-16

ID3v2.3 APIC Frame



Text Encoding is the encoding of the Description field. Its value is either ISO-8859-1 or UCS-16 - the same as in text frames. MIME Type is a NULL-terminated, ASCII C-string which contains the image's MIME type, such as 'image/jpeg' or 'image/png'. Description is a NULL-terminated C-string which contains a text description of the image.

value	type	value	type
0	Other	1	32x32 pixels 'file icon' (PNG only)
2	Other file icon	3	Cover (front)
4	Cover (back)	5	Leaflet page
6	Media (e.g. label side of CD)	7	Lead artist / Lead performer / Soloist
8	Artist / Performer	9	Conductor
10	Band / Orchestra	11	Composer
12	Lyricist / Text writer	13	Recording location
14	During recording	15	During performance
16	Movie / Video screen capture	17	A bright colored fish
18	Illustration	19	Band / Artist logotype
20	Publisher / Studio logotype		

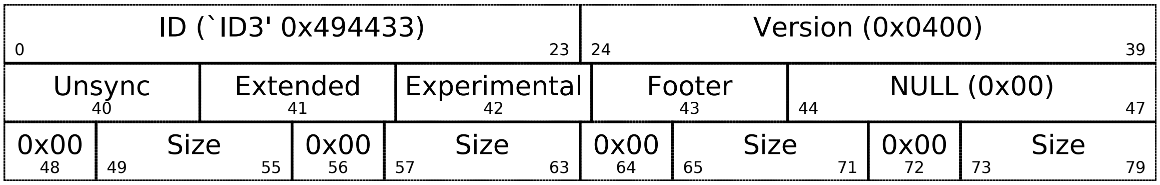
Table 10.3: APIC image types

ID3v2.3 Frame IDs

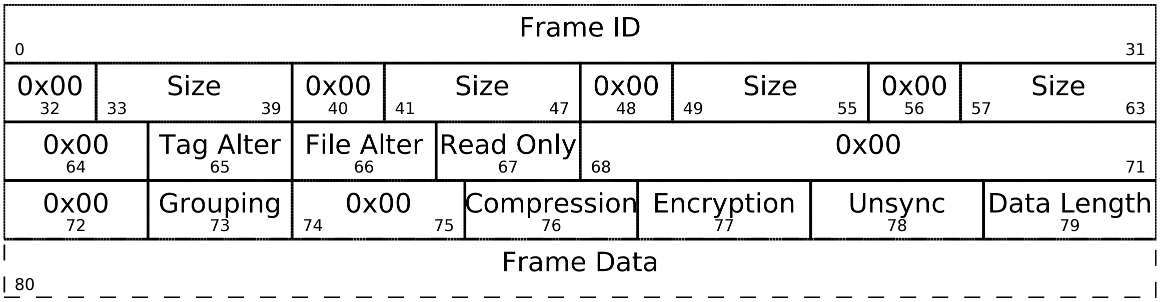
`AENC`	Audio encryption
`APIC`	Attached picture
`COMM`	Comments
`COMR`	Commercial frame
`ENCR`	Encryption method registration
`EQUA`	Equalization
`ETCO`	Event timing codes
`GEOB`	General encapsulated object
`GRID`	Group identification registration
`IPLS`	Involved people list
`LINK`	Linked information
`MCDI`	Music CD identifier
`MLLT`	MPEG location lookup table
`OWNE`	Ownership frame
`PRIV`	Private frame
`PCNT`	Play counter
`POPM`	Popularimeter
`POSS`	Position synchronisation frame
`RBUF`	Recommended buffer size
`RVAD`	Relative volume adjustment
`RVRB`	Reverb
`SYLT`	Synchronized lyric / text
`SYTC`	Synchronized tempo codes
`TALB`	Album /Movie / Show title
`TBPM`	BPM (beats per minute)
`TCOM`	Composer
`TCON`	Content type
`TCOP`	Copyright message
`TDAT`	Date
`TDLY`	Playlist delay
`TENC`	Encoded by
`TEXT`	Lyricist / Text writer
`TFLT`	File type
`TIME`	Time
`TIT1`	Content group description
`TIT2`	Title / songname / content description
`TIT3`	Subtitle / Description refinement
`TKEY`	Initial key
`TLAN`	Language(s)
`TLEN`	Length
`TMED`	Media type
`TOAL`	Original album/movie/show title
`TOFN`	Original filename
`TOLY`	Original lyricist(s) / text writer(s)
`TOPE`	Original artist(s) / performer(s)
`TORY`	Original release year
`TOWN`	File owner / licensee
`TPE1`	Lead performer(s) / Soloist(s)
`TPE2`	Band / orchestra / accompaniment
`TPE3`	Conductor / performer refinement
`TPE4`	Interpreted, remixed, or otherwise modified by
`TPOS`	Part of a set
`TPUB`	Publisher
`TRCK`	Track number / Position in set
`TRDA`	Recording dates
`TRSN`	Internet radio station name
`TRSO`	Internet radio station owner
`TSIZ`	Size
`TSRC`	ISRC (international standard recording code)
`TSSE`	Software/Hardware and encoding settings
`TYER`	Year
`TXXX`	User defined text information frame
`UFID`	Unique file identifier
`USER`	Terms of use
`USLT`	Unsychronized lyric / text transcription
`WCOM`	Commercial information
`WCOP`	Copyright / Legal information
`WOAF`	Official audio file webpage
`WOAR`	Official artist/performer webpage
`WOAS`	Official audio source webpage
`WORS`	Official internet radio station homepage
`WPAY`	Payment
`WPUB`	Publishers official webpage
`WXXX`	User defined URL link frame

10.3.3 ID3v2.4

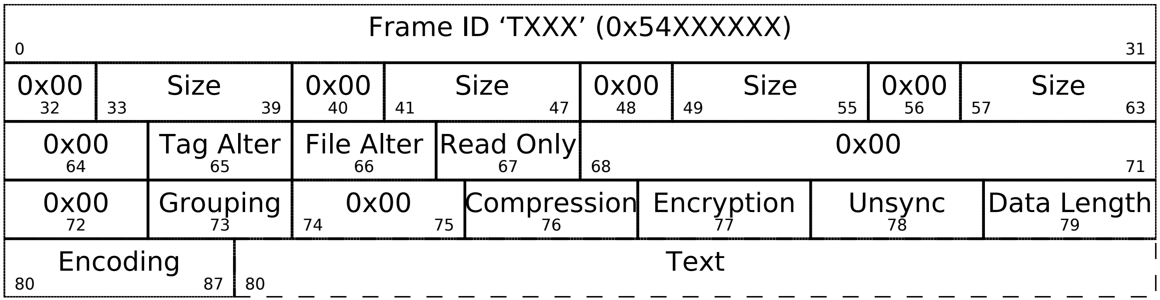
ID3v2.4 Header



ID3v2.4 Frame

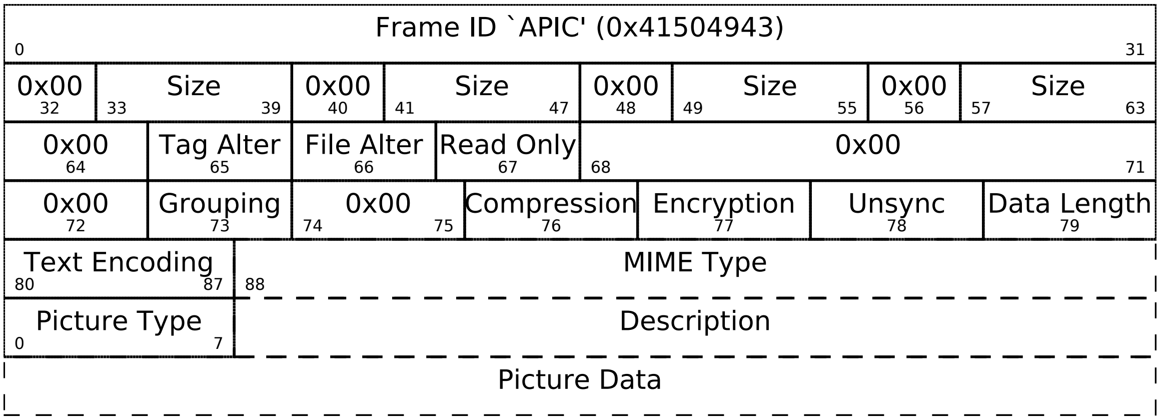


Frame IDs that begin with the letter 'T' (0x54) are text frames. These have an additional text encoding byte before the actual text data. All text strings may be terminated by a null character (0x00 or 0x0000, depending on the encoding).



Encoding Byte	Text Encoding
0x00	ISO-8859-1
0x01	UTF-16
0x02	UTF-16BE
0x03	UTF-8

ID3v2.4 APIC Frame



Text Encoding is the encoding of the Description field. Its value is either ISO-8859-1, UTF-16 or UTF-8 - the same as in text frames. MIME Type is a NULL-terminated, ASCII C-string which contains the image's MIME type, such as 'image/jpeg' or 'image/png'. Description is a NULL-terminated C-string which contains a text description of the image.

value	type	value	type
0	Other	1	32x32 pixels 'file icon' (PNG only)
2	Other file icon	3	Cover (front)
4	Cover (back)	5	Leaflet page
6	Media (e.g. label side of CD)	7	Lead artist / Lead performer / Soloist
8	Artist / Performer	9	Conductor
10	Band / Orchestra	11	Composer
12	Lyricist / Text writer	13	Recording location
14	During recording	15	During performance
16	Movie / Video screen capture	17	A bright colored fish
18	Illustration	19	Band / Artist logotype
20	Publisher / Studio logotype		

Table 10.4: APIC image types

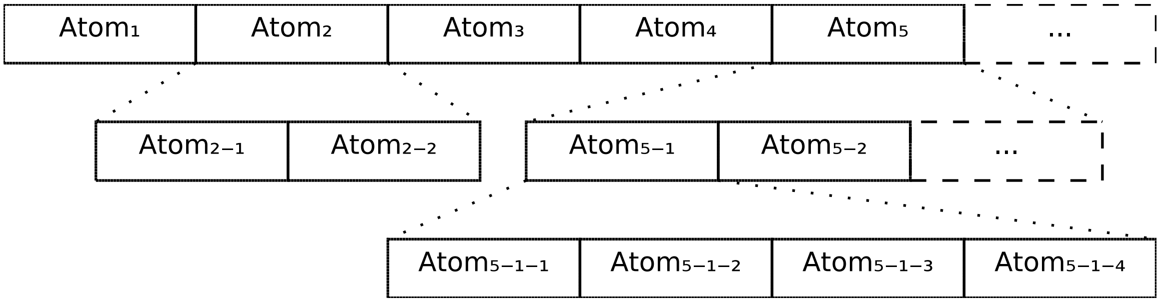
ID3v2.4 Frame IDs

AENC	Audio encryption
APIC	Attached picture
ASPI	Audio seek point index
COMM	Comments
COMR	Commercial frame
ENCR	Encryption method registration
EQU2	Equalisation (2)
ETCO	Event timing codes
GEOB	General encapsulated object
GRID	Group identification registration
LINK	Linked information
MCDI	Music CD identifier
MLLT	MPEG location lookup table
OWNE	Ownership frame
PRIV	Private frame
PCNT	Play counter
POPM	Popularimeter
POSS	Position synchronisation frame
RBUF	Recommended buffer size
RVA2	Relative volume adjustment (2)
RVRB	Reverb
SEEK	Seek frame
SIGN	Signature frame
SYLT	Synchronised lyric/text
SYTC	Synchronised tempo codes
TALB	Album/Movie/Show title
TBPM	BPM (beats per minute)
TCOM	Composer
TCON	Content type
TCOP	Copyright message
TDEN	Encoding time
TDLY	Playlist delay
TDOR	Original release time
TDRC	Recording time
TDRL	Release time
TDTG	Tagging time
TENC	Encoded by
TEXT	Lyricist/Text writer
TFLT	File type
TIPL	Involved people list
TIT1	Content group description
TIT2	Title/songname/content description
TIT3	Subtitle/Description refinement
TKEY	Initial key
TLAN	Language(s)
TLEN	Length
TMCL	Musician credits list
TMED	Media type
TMOO	Mood
TOAL	Original album/movie/show title
TOFN	Original filename
TOLY	Original lyricist(s)/text writer(s)
TOPE	Original artist(s)/performer(s)
TOWN	File owner/licensee
TPE1	Lead performer(s)/Soloist(s)
TPE2	Band/orchestra/accompaniment
TPE3	Conductor/performer refinement
TPE4	Interpreted, remixed, or otherwise modified by
TPOS	Part of a set
TPRO	Produced notice
TPUB	Publisher
TRCK	Track number/Position in set
TRSN	Internet radio station name
TRSO	Internet radio station owner
TSOA	Album sort order
TSOP	Performer sort order
TSOT	Title sort order
TSRC	ISRC (international standard recording code)
TSSE	Software/Hardware and settings used for encoding
TSST	Set subtitle
TXXX	User defined text information frame
UFID	Unique file identifier
USER	Terms of use
USLT	Unsynchronised lyric/text transcription
WCOM	Commercial information
WCOP	Copyright/Legal information
WOAF	Official audio file webpage
WOAR	Official artist/performer webpage
WOAS	Official audio source webpage
WORS	Official Internet radio station homepage
WPAY	Payment
WPUB	Publishers official webpage
WXXX	User defined URL link frame

11 M4A

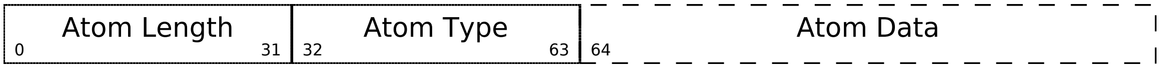
M4A is typically AAC audio in a QuickTime container stream, though it may also contain other formats such as MPEG-1 audio.

11.1 the QuickTime File Stream

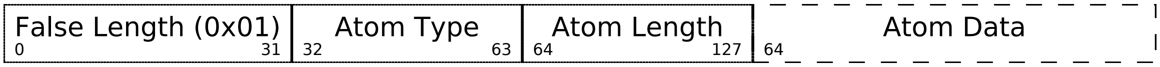


Unlike other chunked formats such as RIFF WAVE, QuickTime's atom chunks may be containers for other atoms. All of its fields are big-endian.

11.1.1 a QuickTime Atom



'Atom Type' is an ASCII string. 'Atom Length' is the length of the entire atom, including the header. If 'Atom Length' is 0, the atom continues until the end of the file. If 'Atom Length' is 1, the atom has an extended size. This means there is a 64-bit length field immediately after the header which is the atom's actual size.

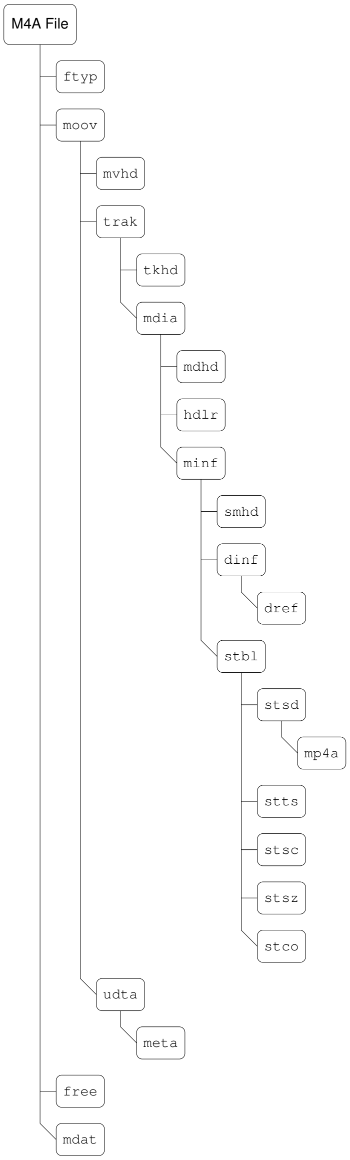


11.1.2 Container Atoms

There is no flag or field to tell a QuickTime parser which of its atoms are containers and which ones are not. If an atom is known to be a container, one can treat its Atom Data as a QuickTime stream and parse it in a recursive fashion.

11.2 M4A Atoms

A typical M4A begins with an 'ftyp' atom indicating its file type, followed by a 'moov' atom containing a copious amount of file metadata, an optional 'free' atom with nothing but empty space (so that metadata can be resized, if necessary) and an 'mdat' atom containing the song data itself.



11.2.1 the ftyp Atom

ftyp Length 0 31	`ftyp' (0x66747970) 32 63
Major Brand 64 95	Major Brand Version 96 127
Compatible Brand$_1$ 128 159	... 160

The 'Major Brand' and 'Compatible Brand' fields are ASCII strings. 'Major Brand Version' is an integer.

11.2.2 the mvhd Atom

mvhd Length 0 31		`mvhd' (0x6D766864) 32 63	
Version 64 71	Flags (0x000000) 72 95		
Created Mac UTC Date 96 127/159		Modified Mac UTC Date 128/160 159/223	
Time Scale 160/224 191/255		Duration 192/256 223/319	
Playback Speed 224/320 255/351		User Volume 256/352 271/367	Reserved (0) 272/368 351/447
WGM A 352/448 383/479	WGM B 384/480 415/511	WGM U 416/512 447/543	WGM C 448/544 479/575
WGM D 480/576 511/607	WGM V 512/608 543/639	WGM X 544/640 575/671	WGM Y 576/672 607/703
WGM W 608/704 639/735	QuickTime Preview 640/736 703/799		
QuickTime Still Poster 704/800 735/831		QuickTime Selection Time 736/832 799/895	
QuickTime Current Time 800/896 831/927		next/new track ID 832/928 863/959	

If 'Version' is 0, 'Created Mac UTC Date', 'Modified Mac UTC Date' and 'Duration' are 32-bit fields. If it is 1, they are 64-bit fields.

11.2.3 the tkhd Atom

tkhd Length 0 31	`tkhd' (0x746B6864) 32 63				
Version 64 71	Reserved (0) 72 91	Track in Poster 92	Track in Preview 93	Track in Movie 94	Track Enabled 95
Created Mac UTC Date 96 127/159	Modified Mac UTC Date 128/160 159/223				
Track ID 160/224 191/255	Reserved (0) 192/256 255/319				
Duration 256/320 287/383	Reserved (0) 288/384 319/415				
Video Layer 320/415 335/431	QuickTime Alt 336/432 351/447	Audio Volume 352/448 367/463	Reserved (0) 368/464 383/479		
VGM value A 384/480 415/511	VGM value B 416/512 447/543	VGM value U 448/544 479/575	VGM value C 480/576 511/607		
VGM value D 512/608 543/639	VGM value V 544/640 575/671	VGM value X 576/672 607/703	VGM value Y 608/704 639/735		
VGM value W 640/736 671/767	Video Width 672/768 703/799	Video Height 704/800 735/831			

As with 'mvhd', if 'Version' is 0, 'Created Mac UTC Date', 'Modified Mac UTC Date' and 'Duration' are 32-bit fields. If it is 1, they are 64-bit fields.

11.2.4 the mdhd Atom

The mdhd atom contains track information such as samples-per-second, track length and creation/modification times.

mdhd Length 0 31	`mdhd' (0x6D646864) 32 63	
Version 64 71	Flags (0x000000) 72 95	
Created Mac UTC Date 96 127/159	Modified Mac UTC Date 128/60 159/223	
Sample Rate 160/224 191/255	Track Length 192/256 223/319	
Pad 224/320	Language 225/321 239/335	Quality 240/336 255/351

As with 'mvhd', if 'Version' is 0, 'Created Mac UTC Date', 'Modified Mac UTC Date' and 'Track Length' are 32-bit fields. If it is 1, they are 64-bit fields.

11.2.5 the hdlr Atom

hdlr Length 0 31	`hdlr' (0x68646C72) 32 63
Version 64 71	Flags (0x000000) 72 95
QuickTime type 96 127	Subtype/media type 128 159
Quicktime manufacturer 160 191	
QuickTime flags 192 223	Quicktime flags mask 224 255
Component Name Length 256 263	Component Name 264

'QuickTime flags', 'QuickTime flags mask' and 'Component Name Length' are integers. The rest are ASCII strings.

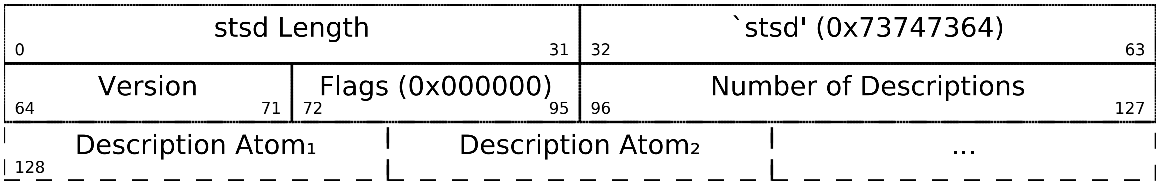
11.2.6 the smhd Atom

smhd Length 0 31		`smhd' (0x736D6864) 32 63	
Version 64 71	Flags (0x000000) 72 95	Audio Balance 96 111	Reserved (0x0000) 112 127

11.2.7 the dref Atom

dref Length 0 31		`dref' (0x64726566) 32 63
Version 64 71	Flags (0x000000) 72 95	Number of References 96 127
Reference Atom₁ 128	Reference Atom₂	...

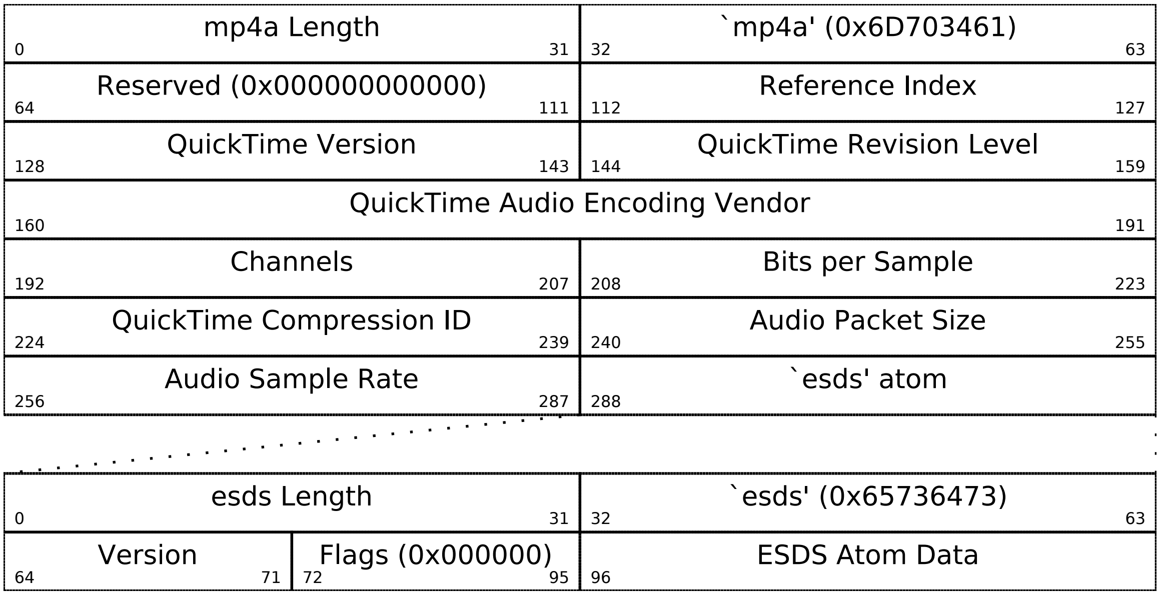
11.2.8 the stsd Atom



stsd Length	0	31
`stsd' (0x73747364)	32	63
Version	64	71
Flags (0x000000)	72	95
Number of Descriptions	96	127
Description Atom₁	128	
Description Atom₂		
...		

11.2.9 the mp4a Atom

The `mp4a` atom contains information such as the number of channels and bits-per-sample. It can be found in the `stsd` atom.



mp4a Length	0	31
`mp4a' (0x6D703461)	32	63
Reserved (0x000000000000)	64	111
Reference Index	112	127
QuickTime Version	128	143
QuickTime Revision Level	144	159
QuickTime Audio Encoding Vendor	160	191
Channels	192	207
Bits per Sample	208	223
QuickTime Compression ID	224	239
Audio Packet Size	240	255
Audio Sample Rate	256	287
`esds' atom	288	

esds Length	0	31
`esds' (0x65736473)	32	63
Version	64	71
Flags (0x000000)	72	95
ESDS Atom Data	96	

11.2.10 the stts Atom

stts Length	0	31
`stts' (0x73747473)	32	63
Version	64	71
Flags (0x000000)	72	95
Number of Times	96	127
Frame Count$_1$	128	159
Duration$_1$	160	191
Frame Count$_2$	192	223
Duration$_2$	224	255
...	256	

11.2.11 the stsc Atom

stsc Length	0	31
`stsc' (0x73747363)	32	63
Version	64	71
Flags (0x000000)	72	95
Number of Blocks	96	127
First Chunk$_1$	128	159
Samples per Chunk$_1$	160	191
Sample Duration Index$_1$	192	223
First Chunk$_2$	224	255
Samples per Chunk$_2$	256	287
Sample Duration Index$_2$	288	319
...	320	

11.2.12 the stsz Atom

stsz Length	0	31
`stsz' (0x7374737A)	32	63
Version	64	71
Flags (0x000000)	72	95
Number of Block Sizes	96	127
Block Size$_1$	128	159
Block Size$_2$	160	191
...	192	

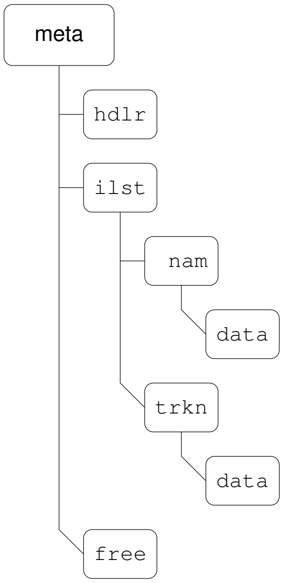
11.2.13 the stco Atom

<table>
<tr><td colspan="12">stco Length
0 31</td><td colspan="12">`stco' (0x7374636F)
32 63</td></tr>
<tr><td colspan="6">Version
64 71</td><td colspan="6">Flags (0x000000)
72 95</td><td colspan="12">Number of Offsets
96 127</td></tr>
<tr><td colspan="8">Offset$_1$
128 159</td><td colspan="8">Offset$_2$
160 191</td><td colspan="8">...
192</td></tr>
</table>

Offsets point to an absolute position in the M4A file of AAC data in the `mdat` atom. Therefore, if the `moov` atom size changes (which can happen by writing new metadata in its `meta` child atom) the `mdat` atom may move and these absolute offsets will change. In that instance, they **must** be re-adjusted in the `stco` atom or the file may become unplayable.

11.2.14 the meta Atom

<table>
<tr><td colspan="2">meta Length
0 31</td><td colspan="2">`meta' (0x6D657461)
32 63</td></tr>
<tr><td>Version
64 71</td><td colspan="3">Flags (0x000000)
72 95</td></tr>
<tr><td>`hdlr' atom<br>96</td><td>`ilst' atom</td><td>`free' atom</td><td>...</td></tr>
</table>



The atoms within the `ilst` container are all containers themselves, each with a `data` atom of its own. Notice that many of `ilst`'s sub-atoms begin with the non-ASCII 0xA9 byte.

<table>
<tr><td>data Length
0 31</td><td>`data' (0x64617461)
32 63</td></tr>
<tr><td>Type
64 95</td><td>Reserved (0x00000000)
96 127</td></tr>
<tr><td colspan="2">Data
128</td></tr>
</table>

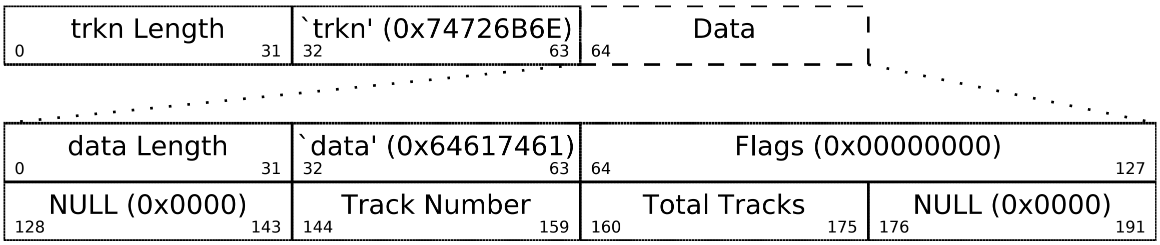
Text data atoms have a 'Type' of 1. Binary data atoms typically have a 'Type' of 0.

Atom	Description	Atom	Description	Atom	Description
`alb`	Album Nam	`ART`	Track Artist	`cmt`	Comments
`covr`	Cover Image	`cpil`	Compilation	`cprt`	Copyright
`day`	Year	`disk`	Disc Number	`gnre`	Genre
`grp`	Grouping	`----`	iTunes-specific	`nam`	Track Name
`rtng`	Rating	`tmpo`	BMP	`too`	Encoder
`trkn`	Track Number	`wrt`	Composer		

Table 11.1: Known `ilst` sub-atoms

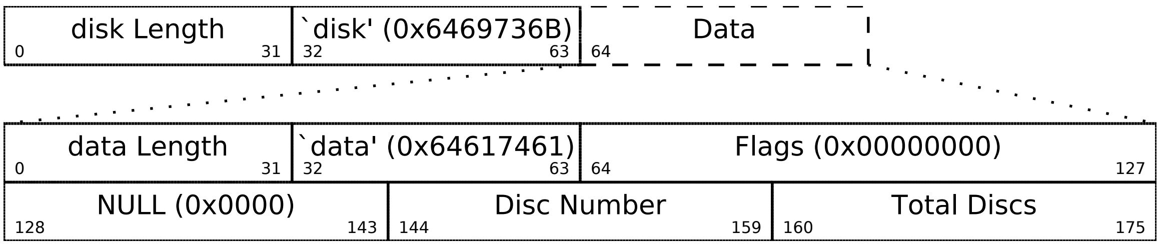
the trkn Sub-Atom

`trkn` is a binary sub-atom of `meta` which contains the track number.



the disk Sub-Atom

`disk` is a binary sub-atom of `meta` which contains the disc number. For example, if the track belongs to the first disc in a set of two discs, the sub-atom will contain that information.



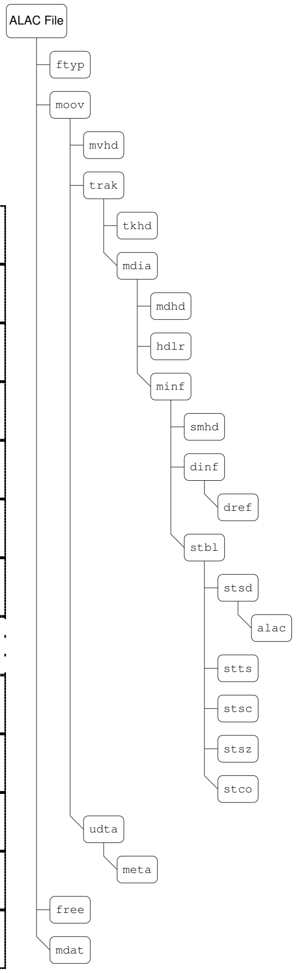
12 Apple Lossless

Apple's lossless audio codec, informally referred to as "ALAC", is lossless audio inside a QuickTime container - similar to M4A. Its stream is the same collection of atoms as covered on page 127. The key difference is the contents of its mdat atom.

12.1 the ALAC File Stream

This is the typical arrangements of ALAC atoms as encoded by iTunes. As you can see, it is almost identical to the layout of AAC audio. One of the key differences is that ALAC's stsd atom contains an alac description sub-atom rather than an mp4a description atom.

Its layout is as follows:

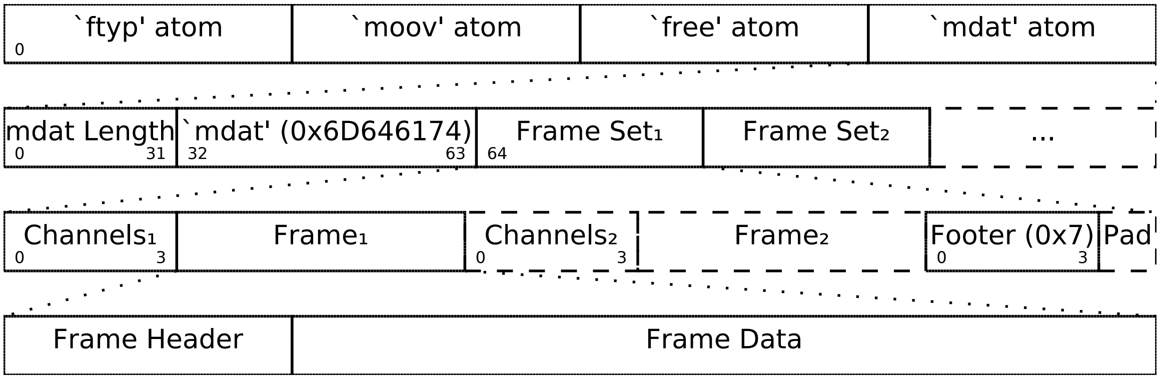


alac Length	0	31
\`alac' (0x616C6163)	32	63
Reserved (0x000000000000)	64	111
Reference Index	112	127
QuickTime Version	128	143
QuickTime Revision Level	144	159
QuickTime Audio Encoding Vendor	160	191
Channels	192	207
Bits per Sample	208	223
QuickTime Compression ID	224	239
Audio Packet Size	240	255
Audio Sample Rate	256	287
\`alac' atom	288	

alac Length	0	31
\`alac' (0x616C6163)	32	63
Padding (0x00000000)	64	95
Max Samples Per Frame	96	127
Padding (0)	128	135
Bits per Sample	136	143
History Multiplier	144	151
Initial History	152	159
Maximum K	160	167
Channels	168	175
Unknown	176	191
Max Coded Frame Size	192	223
Bitrate	224	255
Sample Rate	256	287

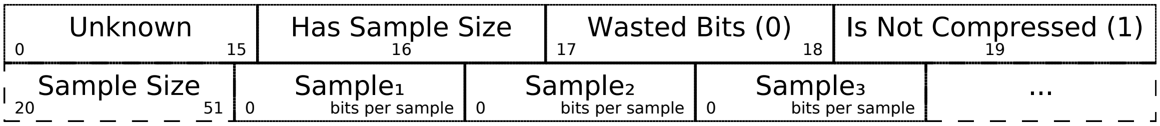
12.2 ALAC Decoding

An ALAC stream is made up of individual framesets within the `mdat` atom, as follows:



Note that the channels field is the total number of channels, offset by 1; `0` is mono and `1` is stereo. The frames within a frameset continue until the value `0x7` is encountered in the channel field. Each frameset will contain only a single frame in most cases. However, multichannel audio may contain more. For example, a 6 channel stream may have framesets with 4 frames each where Frame$_1$ has channels 1-2, Frame$_2$ has channel 3, Frame$_3$ has channel 4 and Frame$_4$ has channel 5-6.

ALAC frames come in two varieties: compressed and uncompressed, depending on the 'Is Not Compressed' bit in the frame header. An uncompressed frame is laid out as follows:

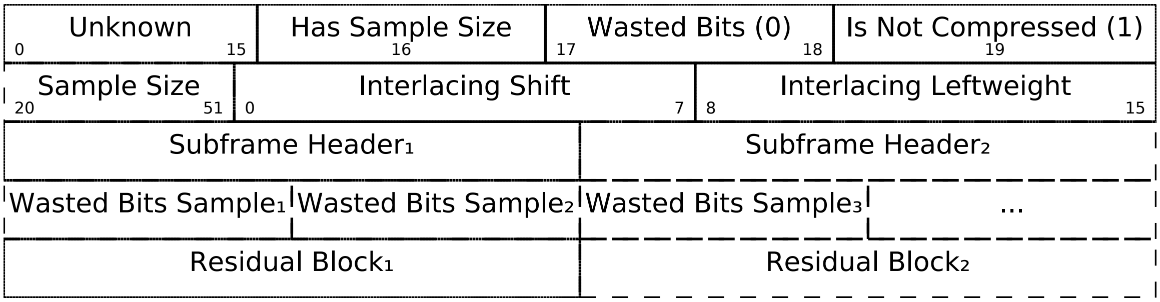


The number of PCM frames in an ALAC frame depends on the 'Has Sample Size' bit. If set, the number of PCM frames equals the 32-bit 'Sample Size' value. If not set, the total number of PCM frames equals the 'Max Coded Frame Size' value in the `alac` atom and the 'Sample Size' value is omitted. ALAC streams typically use the same number of samples per frame until the end of the stream, at which point the leftover samples are placed in a different-sized frame.

Uncompressed frames interleave samples between channels during decoding. For example, a 2 channel frame places Sample$_1$ on Channel$_1$, Sample$_2$ on Channel$_2$, Sample$_3$ on Channel$_1$, Sample$_4$ on Channel$_2$ and so on.

Finally, note that all ALAC framesets have padding as needed such that each new frameset begins on an aligned byte boundary.

A compressed frame is laid out as follows:

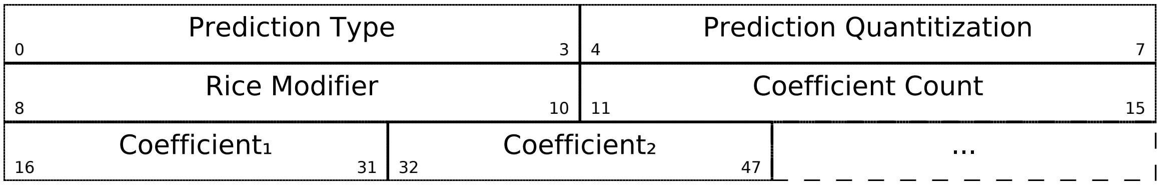


'Interlacing Shift' and 'Interlacing Leftweight' are used for channel decorrelation after the subframes have been decoded.

There is one subframe header and one residual block per channel. If 'Wasted Bits' is greater than 0, there is also a block of wasted bits samples after the subframe headers but before the residuals.

'Wasted Bits' is an attempt to store the least significant bits of each sample more efficiently at high bits-per-sample, where that data will often be indistinguishable from random noise. In effect, it's a block of interlaced uncompressed samples (similar to an uncompressed ALAC frame) each 8 × 'Wasted Bits' bits large. Those wasted bits are then prepended to each sample after channel decorrelation. This process is explained in more detail on page 144.

Each subframe header is laid out as follows:



There are 'Coefficient Count' number of coefficients in each subframe header, each a 16-bit signed value.

12.2.1 Residual Decoding

There are ‘Sample Size’ number of residuals per residual block. Decoding a residual block requires knowing the ‘Initial History’, ‘History Multiplier’, ‘Maximum K’, ‘Channels’, ‘Wasted Bits’ and ‘Bits per Sample’ values, from the `alac` atom and frame header. Fortunately, most of these values are rarely used; we’ll mostly be concerned with the ‘History’ and ‘History Multiplier’.

For each residual block, ‘History’ starts with the value of ‘Initial History’ and will change during residual decoding. We use it to calculate κ using the following formula:

$$\kappa = \left\lfloor \log_2 \left(\frac{\text{history}}{2^9} + 3 \right) \right\rfloor \tag{12.1}$$

Note that if κ exceeds the ‘Maximum K’ value from the `alac` atom, ‘Maximum K’ is used instead.

We then need to know the ‘Bits per Sample’ value of the residual block, which is equal to:

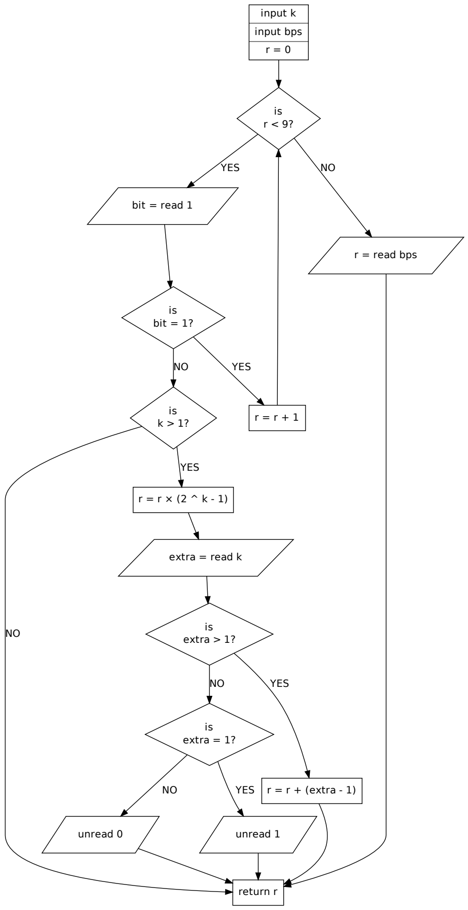
$$\text{Bits per Sample}_{Residual} = \text{Bits per Sample}_{ALAC} - (\text{Wasted Bits} \times 8) + \text{Channels} - 1 \tag{12.2}$$

For mono streams, this is typically equal to the stream’s ‘Bits per Sample’, while for stereo streams it’s often the ‘Bits per Sample’ plus 1.

The κ and ‘Bits per Sample’ values are used to read a single unsigned residual value in the following way:

The initial bit reading portion of the process involves reading a unary value with a stop bit of ‘0’ and a maximum value of 8. If the maximum value is exceeded, we read ‘Bits per Sample’ number of bits as our final value (the only place ‘Bits per Sample’ is used throughout the residual decoding process).

Otherwise, we read κ number of ‘extra’ bits if $\kappa > 1$. If ‘extra’ is greater than 1, we return (‘unary’ $\times$ $(2^k - 1)) +$ (‘extra’ $- 1$). If not, we push a single ‘extra’ bit back on the stream and return our ‘unary’ $\times$ $(2^k - 1)$ value.



We perform the following to convert unsigned residuals to signed residuals which are returned to the subframe decoder:

$$\text{signed} = \begin{cases} \lfloor(\text{unsigned} + 1) \div 2\rfloor & \text{if unsigned is even} \\ -\lfloor(\text{unsigned} + 1) \div 2\rfloor & \text{if unsigned is odd} \end{cases} \tag{12.3}$$

Finally, we use our unsigned value to update 'history' before reading the next residual:

$$\text{history} = \begin{cases} \text{history} + (\text{unsigned} \times \text{history multiplier}) - \lfloor \frac{\text{history} \times \text{history multiplier}}{2^9} \rfloor & \text{if unsigned} \leq 65535 \\ 65535 & \text{if unsigned} > 65535 \end{cases}$$

Thus far, residual decoding isn't overly complex. We simply calculate κ from 'history', read a residual, update 'history' from its unsigned value and repeat the process until we've read an entire subframe's worth of residuals. The bulk of an ALAC file's residuals will be read in this way. However, ALAC also features an "escape code" for large chunks of 0 residuals (which may happen during a long stretch of digital silence, for example).

If 'history' ever falls below 128, this special case is triggered. First, we read a special 'block size' residual value with a 'Bits per Sample' value of 16 and a κ[1] value of:

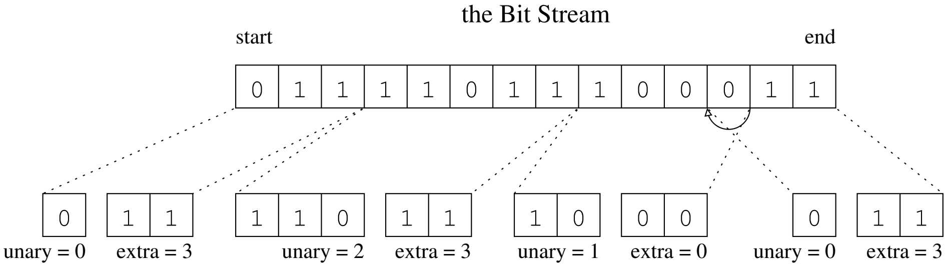
$$\kappa_{\text{blocksize}} = 7 - \log_2(\text{history}) + \frac{\text{history} + 16}{64} \tag{12.4}$$

This 'block size' is how many 0 residuals to send outright - which may be 0, indicating no 0 residuals to send. Either way, if 'block size' $\leq$ 65535, we add 1 to the unsigned value of the next residual in the block (if any). Finally, 'history' is automatically set to 0.

[1] Again, κ cannot exceed 'Maximum K' as encoded in the `alac` atom.

Residual Decoding Example

In this example, we'll decode a group of residuals in which our 'Initial History' is 1130 and our 'History Multiplier' is 40. As a spoiler, κ (the amount of non-unary bits to read after each unary value) will remain 2 for this batch of residuals, but why that is so will be explained below.



- Residual 1
  - $\kappa = \lfloor \log_2((1130 \div 2^9) + 3) \rfloor = \lfloor \log_2(5) \rfloor = 2$
  - $\text{unsigned}_1 = (\mathbf{0} \times (2^2 - 1)) + (\mathbf{3} - 1) = \mathbf{2}$
  - $\text{residual}_1 = \lfloor (2 + 1) \div 2 \rfloor = 1$
  - $\text{history} = 1130 + (\mathbf{2} \times 40) - \lfloor (1130 \times 40) \div 2^9 \rfloor = 1130 + 80 - 88 = 1122$
- Residual 2
  - $\kappa = \lfloor \log_2((1122 \div 2^9) + 3) \rfloor = \lfloor \log_2(5) \rfloor = 2$
  - $\text{unsigned}_2 = (\mathbf{2} \times (2^2 - 1)) + (\mathbf{3} - 1) = \mathbf{8}$
  - $\text{residual}_2 = \lfloor (8 + 1) \div 2 \rfloor = 4$
  - $\text{history} = 1122 + (\mathbf{8} \times 40) - \lfloor (1122 \times 40) \div 2^9 \rfloor = 1122 + 320 - 87 = 1355$
- Residual 3
  - $\kappa = \lfloor \log_2((1355 \div 2^9) + 3) \rfloor = \lfloor \log_2(5) \rfloor = 2$
  - $\text{unsigned}_3 = (\mathbf{1} \times (2^2 - 1)) = \mathbf{3}$ (note that we "unread" one 0 bit here)
  - $\text{residual}_3 = -\lfloor (3 + 1) \div 2 \rfloor = -2$
  - $\text{history} = 1355 + (\mathbf{3} \times 40) - \lfloor (1355 \times 40) \div 2^9 \rfloor = 1355 + 120 - 105 = 1370$
- Residual 4
  - $\kappa = \lfloor \log_2((1370 \div 2^9) + 3) \rfloor = \lfloor \log_2(5) \rfloor = 2$
  - $\text{unsigned}_4 = (\mathbf{0} \times (2^2 - 1)) + (\mathbf{3} - 1) = \mathbf{2}$
  - $\text{residual}_4 = \lfloor (2 + 1) \div 2 \rfloor = 1$
  - $\text{history} = 1370 + (\mathbf{2} \times 40) - \lfloor (1370 \times 40) \div 2^9 \rfloor = 1370 + 80 - 107 = 1343$

Thus, our batch of signed residual values are 1, 4, -2 and 1.

12.2.2 Subframe Calculation

Given our list of decoded residual values; along with a list of coefficients, a 'Coefficient Count' and a 'Prediction Quantitization' value (all from the subframe header), we can now generate a list of signed subframe samples for a given channel.

The first residual is always the first output sample:

$$\text{Sample}_0 = \text{Residual}_0$$

Then, for the next 'Coefficient Count' number of residuals:

$$\text{Sample}_i = \text{Residual}_i + \text{Sample}_{i-1}$$

For example, given that we have a 'Coefficient Count' of 4 and our first five residuals are 16, 1, 9, -1 and -1; our first five sample values are:

$$\begin{aligned}
\text{Sample}_0 &= \mathbf{16} \\
\text{Sample}_1 &= 16 + 1 = \mathbf{17} \\
\text{Sample}_2 &= 17 + 9 = \mathbf{26} \\
\text{Sample}_3 &= 26 - 1 = \mathbf{25} \\
\text{Sample}_4 &= 25 - 1 = \mathbf{24}
\end{aligned}$$

These are our "starting point" samples upon which the remainder of the subframe will be built.

Subsequent samples are calculated in the following way:

$$\begin{aligned}
\text{LPC Sum}_i &= \sum_{j=0}^{coeffs-1} \text{Coefficient}_j \times (\text{Sample}_{i-j-1} - \text{Sample}_{i-coeffs-1}) \\
\text{Sample}_i &= \left\lfloor \frac{\text{LPC Sum}_i + 2^{\text{Predictor Quantitization - 1}}}{2^{\text{Predictor Quantitization}}} \right\rfloor + \text{Residual}_i + \text{Sample}_{i-coeffs-1}
\end{aligned}$$

For example, given $\text{Residual}_5 = 1$, 'Predictor Quantitization' = 9 and the coefficients 1122, -766, 107 and 122:

$$\begin{aligned}
\text{LPC Sum}_5 &= (1122 \times (24 - 16)) + (-766 \times (25 - 16)) + (107 \times (26 - 16)) + (122 \times (17 - 16)) \\
&= (1122 \times 8) + (-766 \times 9) + (107 \times 10) + (122 \times 1) \\
&= 8976 + -6894 + 1070 + 122 = \mathbf{3274} \\
\text{Sample}_5 &= \left\lfloor \frac{\mathbf{3274} + 2^8}{2^9} \right\rfloor + 1 + 16 = \left\lfloor \frac{3530}{512} \right\rfloor + 1 + 16 = \mathbf{23}
\end{aligned}$$

But before calculating Sample_6, we need to adjust our coefficient list.

Updating the coefficient list first requires the 'sign' function:

$$\operatorname{sign}(x) = \begin{cases} 1 & \text{if } x > 0 \\ 0 & \text{if } x = 0 \\ -1 & \text{if } x < 0 \end{cases} \tag{12.5}$$

We then take our 'samples', index i, 'coefficients', coefficient 'count', 'residual', 'predictor quantitization' values and go through the following routine if residual $\neq 0$:

```
Step 1.  $\text{original sign} = \operatorname{sign}(residual)$
Step 2.  $j = 0$
Step 3.  $v = \text{Sample}_{i-\text{count}-1} - \text{Sample}_{i-\text{count}+j}$
Step 4.  $sign = \begin{cases} \operatorname{sign}(v) & \text{if original sign} > 0 \\ -\operatorname{sign}(v) & \text{if original sign} \leq 0 \end{cases}$
Step 5.  $\text{Coefficient}_{\text{count}-j-1} = \text{Coefficient}_{\text{count}-j-1} - sign$
Step 6.  $residual = residual - \left(\left\lfloor \frac{v \times sign}{2^{\text{Predictor Quantitization}}} \right\rfloor \times (j+1)\right)$
Step 7.  $j = j + 1$
Step 8.  if ($j < count$) and ($\operatorname{sign}(residual) = \text{original sign}$), goto step 3
```

To continue our example:

$$\begin{aligned}
\text{original sign} &= \operatorname{sign}(1) = \mathbf{1} \\
j &= 0 \\
v &= \text{Sample}_0 - \text{Sample}_1 = 16 - 17 = \mathbf{-1} \\
sign &= \operatorname{sign}(-1) = \mathbf{-1} \\
\text{Coefficient}_3 &= \text{Coefficient}_3 - -1 = 122 + 1 = \mathbf{123} \\
residual &= residual - \left(\left\lfloor \frac{-1 \times -1}{2^9} \right\rfloor \times (0+1)\right) = 1 - (0 \times 1) = \mathbf{1} \\
j &= 1 \\
v &= \text{Sample}_0 - \text{Sample}_2 = 16 - 26 = \mathbf{-10} \\
sign &= \operatorname{sign}(-10) = \mathbf{-1} \\
\text{Coefficient}_2 &= \text{Coefficient}_2 - -1 = 107 + 1 = \mathbf{108} \\
residual &= residual - \left(\left\lfloor \frac{-10 \times -1}{2^9} \right\rfloor \times (1+1)\right) = 1 - (0 \times 2) = \mathbf{1} \\
j &= 2 \\
v &= \text{Sample}_0 - \text{Sample}_3 = 16 - 25 = \mathbf{-9} \\
sign &= \operatorname{sign}(-9) = \mathbf{-1} \\
\text{Coefficient}_1 &= \text{Coefficient}_1 - -1 = -766 + 1 = \mathbf{-765} \\
residual &= residual - \left(\left\lfloor \frac{-9 \times -1}{2^9} \right\rfloor \times (2+1)\right) = 1 - (0 \times 3) = \mathbf{1} \\
j &= 3 \\
v &= \text{Sample}_0 - \text{Sample}_4 = 16 - 25 = \mathbf{-8} \\
sign &= \operatorname{sign}(-8) = \mathbf{-1} \\
\text{Coefficient}_0 &= \text{Coefficient}_0 - -1 = 1122 + 1 = \mathbf{1123} \\
residual &= residual - \left(\left\lfloor \frac{-8 \times -1}{2^9} \right\rfloor \times (3+1)\right) = 1 - (0 \times 4) = \mathbf{1} \\
j &= 4 \text{ and stop, since 4 equals our coefficient count}
\end{aligned}$$

Given that Residual_6 is -4, the calculation of Sample_6 is as follows:

$$\begin{aligned}
\text{LPC Sum}_6 &= (1123 \times (23 - 17)) + (-765 \times (24 - 17)) + (108 \times (25 - 17)) + (123 \times (26 - 17)) \\
&= (1123 \times 6) + (-765 \times 7) + (108 \times 8) + (123 \times 9) \\
&= 6738 + -5355 + 864 + 1107 = \mathbf{3354} \\
\text{Sample}_6 &= \left\lfloor \frac{\mathbf{3354} + 2^8}{2^9} \right\rfloor + -4 + 17 = \left\lfloor \frac{3610}{512} \right\rfloor + -4 + 17 = \mathbf{20} \\
\text{original sign} &= \text{sign}(-4) = -1 \\
j &= 0 \\
v &= \text{Sample}_1 - \text{Sample}_2 = 17 - 26 = \mathbf{\text{-}9} \\
\textit{sign} &= -\text{sign}(-9) = \mathbf{1} \\
\text{Coefficient}_3 &= \text{Coefficient}_3 - 1 = 123 - 1 = \mathbf{122} \\
\textit{residual} &= \textit{residual} - \left(\left\lfloor \frac{-9 \times 1}{2^9} \right\rfloor \times (0 + 1) \right) = -4 - (-1 \times 1) = \mathbf{\text{-}3} \\
j &= 1 \\
v &= \text{Sample}_1 - \text{Sample}_3 = 17 - 25 = \mathbf{\text{-}8} \\
\textit{sign} &= -\text{sign}(-8) = \mathbf{1} \\
\text{Coefficient}_2 &= \text{Coefficient}_2 - 1 = 108 - 1 = \mathbf{107} \\
\textit{residual} &= \textit{residual} - \left(\left\lfloor \frac{-8 \times 1}{2^9} \right\rfloor \times (1 + 1) \right) = -3 - (-1 \times 2) = \mathbf{\text{-}1} \\
j &= 2 \\
v &= \text{Sample}_1 - \text{Sample}_4 = 17 - 24 = \mathbf{\text{-}7} \\
\textit{sign} &= -\text{sign}(-7) = \mathbf{1} \\
\text{Coefficient}_1 &= \text{Coefficient}_1 - 1 = -765 - 1 = \mathbf{\text{-}766} \\
\textit{residual} &= \textit{residual} - \left(\left\lfloor \frac{-7 \times 1}{2^9} \right\rfloor \times (2 + 1) \right) = -1 - (-1 \times 3) = \mathbf{2} \\
&\text{and stop, since sign}(2) \neq \text{sign}(-4) \text{ (our `original sign' value)}
\end{aligned}$$

So, the coefficients for Sample_7 are 1123, -766, 107 and 122.

12.2.3 Channel Decorrelation

If we have more than one channel of output, the next step is performing channel decorrelation. If our 'Interlacing Leftweight' value from the frame header is 0, our channels are stored independently. In that case, the Channel_1 is our left samples and Channel_2 is our right samples.

If 'Interlacing Leftweight' is greater than zero, we calculate samples as follows:

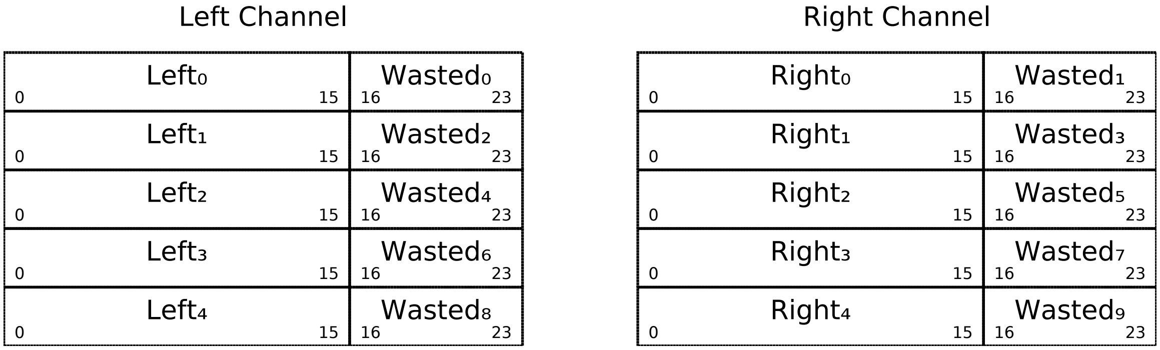
$$\text{Right}_i = \text{Channel}_1 - \left\lfloor \frac{\text{Channel}_2 \times \text{Interlacing Leftweight}}{2^{\text{Interlacing Shift}}} \right\rfloor$$
$$\text{Left}_i = \text{Channel}_2 + \text{Right}_i$$

For example, given the Channel_1 samples of 14, 15, 19, 17, 18; the Channel_2 samples of 16, 17, 26, 25, 24, an 'Interlacing Shift' value of 2 and an 'Interlacing Leftweight' values of 3, we calculate output samples as follows:

Sample	Channel_1	Channel_2	Right_i	Left_i
0	14	16	$14 - \lfloor(16 \times 3) \div 2^2\rfloor = \mathbf{2}$	$16 + \mathbf{2} = \mathbf{18}$
1	15	17	$15 - \lfloor(17 \times 3) \div 2^2\rfloor = \mathbf{3}$	$17 + \mathbf{3} = \mathbf{20}$
2	19	26	$19 - \lfloor(26 \times 3) \div 2^2\rfloor = \mathbf{0}$	$26 + \mathbf{0} = \mathbf{26}$
3	17	25	$17 - \lfloor(25 \times 3) \div 2^2\rfloor = \mathbf{-1}$	$25 + \mathbf{-1} = \mathbf{24}$
4	18	24	$18 - \lfloor(24 \times 3) \div 2^2\rfloor = \mathbf{0}$	$24 + \mathbf{0} = \mathbf{24}$

12.2.4 Wasted Bits

A compressed ALAC frame with 'Wasted Bits' stores them interleaved between channels. Then, after channel decorrelation, these verbatim values are prepended to each sample. For example, given a 2 channel stream with 24 bits-per-sample and a 'Wasted Bits' value of 1 (meaning our "wasted" samples are 8 bits large), our final output is as follows:



12.3 ALAC Encoding

To encode an ALAC file, we need a stream of PCM sample integers along with that stream's sample rate, bits-per-sample and number of channels. We'll start by encoding all of the non-audio ALAC atoms, most of which are contained within the `moov` atom. There's over twenty atoms in a typical ALAC file, most of which are packed with seemingly redundant or nonessential data, so it will take awhile before we can move on to the actual audio encoding process.

Remember, all of an ALAC's fields are big-endian.

12.3.1 ALAC Atoms

We'll encode our ALAC file in iTunes order, which means it contains the `ftyp`, `moov`, `free` and `mdat` atoms, in that order.

the ftyp Atom

Field	Size	Value
atom length	32	32
atom type	32	'ftyp' (`0x66747970`)
major brand	32	'M4A ' (`0x4d344120`)
major brand version	32	`0`
compatible brand	32	'M4A ' (`0x4d344120`)
compatible brand	32	'mp42' (`0x6d703432`)
compatible brand	32	'isom' (`0x69736f6d`)
compatible brand	32	`0x00000000`

the moov Atom

Field	Size	Value
atom length	32	`mvhd` size + `trak` size + `udta` size + 8
atom type	32	'moov' (`0x6d6f6f76`)
`mvhd` atom	`mvhd` size	`mvhd` data
`trak` atom	`trak` size	`trak` data
`udta` atom	`udta` size	`udta` data

the mvhd Atom

Field	Size	Value
atom length	32	108/120
atom type	32	'mvhd' (0x6d766864)
version	8	0x00
flags	24	0x000000
created date	32/64	creation date as Mac UTC
modified date	32/64	modification date as Mac UTC
time scale	32	sample rate
duration	32/64	total PCM frames
playback speed	32	0x10000
user volume	16	0x100
padding	80	0x00000000000000000000
window geometry matrix a	32	0x10000
window geometry matrix b	32	0
window geometry matrix u	32	0
window geometry matrix c	32	0
window geometry matrix d	32	0x10000
window geometry matrix v	32	0
window geometry matrix x	32	0
window geometry matrix y	32	0
window geometry matrix w	32	0x40000000
QuickTime preview	64	0
QuickTime still poster	32	0
QuickTime selection time	64	0
QuickTime current time	32	0
next track ID	32	2

If 'version' is 0, 'created date', 'modified date' and 'duration' are 32 bit fields. Otherwise, they are 64 bit fields. The 'created date' and 'modified date' are seconds since the Macintosh Epoch, which is 00:00:00, January 1st, 1904.[2] To convert a Unix Epoch timestamp (seconds since January 1st, 1970) to a Macintosh Epoch, one needs to add 24,107 days - or 2082844800 seconds.

[2] Why 1904? It's the first leap year of the 20th century.

the trak Atom

Field	Size	Value
atom length	32	`tkhd` size + `mdia` size + 8
atom type	32	‘trak’ (`0x7472616b`)
`tkhd` atom	`tkhd` size	`tkhd` data
`mdia` atom	`mdia` size	`mdia` data

the tkhd Atom

Field	Size	Value
atom length	32	92/104
atom type	32	‘tkhd’ (`0x746b6864`)
version	8	`0x00`
padding	20	`0x000000`
track in poster	1	`0`
track in preview	1	`1`
track in movie	1	`1`
track enabled	1	`1`
created date	32/64	creation date as Mac UTC
modified date	32/64	modification date as Mac UTC
track ID	32	`1`
padding	32	`0x00000000`
duration	32/64	total PCM frames
padding	64	`0x0000000000000000`
video layer	16	`0`
QuickTime alternate	16	`0`
volume	16	`0x1000`
padding	16	`0x0000`
video geometry matrix a	32	`0x10000`
video geometry matrix b	32	`0`
video geometry matrix u	32	`0`
video geometry matrix c	32	`0`
video geometry matrix d	32	`0x10000`
video geometry matrix v	32	`0`
video geometry matrix x	32	`0`
video geometry matrix y	32	`0`
video geometry matrix w	32	`0x40000000`
video width	32	`0`
video height	32	`0`

the mdia Atom

Field	Size	Value
atom length	32	`mdhd` size + `hdlr` size + `minf` size + 8
atom type	32	'mdia' (`0x6d646961`)
`mdhd` atom	`mdhd` size	`mdhd` data
`hdlr` atom	`hdlr` size	`hdlr` data
`minf` atom	`minf` size	`minf` data

the mdhd Atom

Field	Size	Value
atom length	32	32/44
atom type	32	'mdhd' (`0x6d646864`)
version	8	`0x00`
flags	24	`0x000000`
created date	32/64	creation date as Mac UTC
modified date	32/64	modification date as Mac UTC
time scale	32	sample rate
duration	32/64	total PCM frames
padding	1	`0`
language	5	
language	5	language value as ISO 639-2
language	5	
QuickTime quality	16	`0`

Note the three, 5-bit 'language' fields. By adding 0x60 to each value and converting the result to ASCII characters, the result is an ISO 639-2 string of the file's language representation. For example, given the values `0x15`, `0x0E` and `0x04`:

$$\begin{aligned}\text{language}_0 &= \texttt{0x15} + \texttt{0x60} = \texttt{0x75} = \texttt{u}\\ \text{language}_1 &= \texttt{0x0E} + \texttt{0x60} = \texttt{0x6E} = \texttt{n}\\ \text{language}_2 &= \texttt{0x04} + \texttt{0x60} = \texttt{0x64} = \texttt{d}\end{aligned}$$

Which is the code '`und`', meaning 'undetermined' - which is typical.

the hdlr Atom

Field	Size	Value
atom length	32	33 + component
atom type	32	‘hdlr’ (0x68646c72)
version	8	0x00
flags	24	0x000000
QuickTime type	32	0x00000000
QuickTime subtype	32	‘soun’ (0x736f756e)
QuickTime manufacturer	32	0x00000000
QuickTime component reserved flags	32	0x00000000
QuickTime component reserved flags mask	32	0x00000000
component name length	8	0x00
component name	component name length × 8	

the minf Atom

Field	Size	Value
atom length	32	`smhd` size + `dinf` size + `stbl` size + 8
atom type	32	‘minf’ (0x6d696e66)
`smhd` atom	`smhd` size	`smhd` data
`dinf` atom	`dinf` size	`dinf` data
`stbl` atom	`stbl` size	`stbl` data

the smhd Atom

Field	Size	Value
atom length	32	16
atom type	32	‘smhd’ (0x736d6864)
version	8	0x00
flags	24	0x000000
audio balance	16	0x0000
padding	16	0x0000

the dinf Atom

Field	Size	Value
atom length	32	`dref` size + 8
atom type	32	‘dinf’ (0x64696e66)
`dref` atom	`dref` size	`dref` data

the dref Atom

Field	Size	Value
atom length	32	28
atom type	32	'dref' (`0x64726566`)
version	8	`0x00`
flags	24	`0x000000`
number of references	32	`1`
reference atom size	32	`12`
reference atom type	32	'url ' (`0x75726c20`)
reference atom data	32	`0x00000001`

the stbl Atom

Field	Size	Value
atom length	32	`stsd` size + `stts` size + `stsc` size + `stsz` size + `stco` size + 8
atom type	32	'stbl' (`0x7374626c`)
`stsd` atom	`stsd` size	`stsd` data
`stts` atom	`stts` size	`stts` data
`stsc` atom	`stsc` size	`stsc` data
`stsz` atom	`stsz` size	`stsz` data
`stco` atom	`stco` size	`stco` data

the stsd Atom

Field	Size	Value
atom length	32	`alac` size + 16
atom type	32	'stsd' (`0x73747364`)
version	8	`0x00`
flags	24	`0x000000`
number of descriptions	32	`1`
`alac` atom	`alac` size	`alac` data

the alac Atom

Field	Size	Value
atom length	32	`72`
atom type	32	'alac' (`0x616c6163`)
reserved	48	`0x000000000000`
reference index	16	`1`
version	16	`0`
revision level	16	`0`
vendor	32	`0x00000000`
channels	16	channel count
bits per sample	16	bits per sample
compression ID	16	`0`
audio packet size	16	`0`
sample rate	16	sample rate
padding	16	`0x0000`
atom length	32	`36`
atom type	32	'alac' (`0x616c6163`)
padding	32	`0x00000000`
max samples per frame	32	largest number of PCM frames per ALAC frame
padding	8	`0x00`
sample size	8	bits per sample
history multiplier	8	`40`
initial history	8	`10`
maximum K	8	`14`
channels	8	channel count
unknown	16	`0x00FF`
max coded frame size	32	largest ALAC frame size, in bytes
bitrate	32	((`mdat` size × 8) ÷ (total PCM frames ÷ sample rate))
sample rate	32	sample rate

The 'history multiplier', 'initial history' and 'maximum K' values are encode-time options, typically set to 40, 10 and 14, respectively.

Note that the 'bitrate' field can't be known in advance; we must fill that value with 0 for now and then return to this atom once encoding is completed and its size has been determined.

the stts Atom

Field	Size	Value
atom length	32	number of times × 8 + 16
atom type	32	‘stts’ (0x73747473)
version	8	0x00
flags	24	0x000000
number of times	32	
frame count 1	32	number of occurrences
frame duration 1	32	PCM frame count
...		

This atom keeps track of how many different sizes of ALAC frames occur in the ALAC file, in PCM frames. It will typically have only two “times”, the block size we’re using for most of our samples and the final block size for any remaining samples.

For example, let’s imagine encoding a 1 minute audio file at 44100Hz with a block size of 4096 frames. This file has a total of 2,646,000 PCM frames (60 × 44100 = 2646000). 2,646,000 PCM frames divided by a 4096 block size means we have 645 ALAC frames of size 4096, and 1 ALAC frame of size 4080.

Therefore:

$$\begin{aligned} \text{number of times} &= 2 \\ \text{frame count}_1 &= 645 \\ \text{frame duration}_1 &= 4096 \\ \text{frame count}_2 &= 1 \\ \text{frame duration}_2 &= 4080 \end{aligned}$$

the stsc Atom

Field	Size	Value
atom length	32	entries × 12 + 16
atom type	32	‘stsc’ (0x73747363)
version	8	0x00
flags	24	0x000000
number of entries	32	
first chunk	32	
ALAC frames per chunk	32	
description index	32	1
...		

This atom stores how many ALAC frames are in a given "chunk". In this instance a "chunk" represents an entry in the `stco` atom table, used for seeking backwards and forwards through the file. 'First chunk' is the starting offset of its frames-per-chunk value, beginning at 1.

As an example, let's take a one minute, 44100Hz audio file that's been broken into 130 chunks (each with an entry in the `stco` atom). Its `stsc` entries would typically be:

$$\begin{aligned}
\text{first chunk}_1 &= 1\\
\text{frames per chunk}_1 &= 5\\
\text{first chunk}_2 &= 130\\
\text{frames per chunk}_2 &= 1
\end{aligned}$$

What this means is that chunks 1 through 129 have 5 ALAC frames each while chunk 130 has 1 ALAC frame. This is a total of 646 ALAC frames, which matches the contents of the `stts` atom.

the stsz Atom

Field	Size	Value
atom length	32	sizes × 4 + 20
atom type	32	'stsz' (`0x7374737a`)
version	8	`0x00`
flags	24	`0x000000`
block byte size	32	`0x00000000`
number of sizes	32	
frame size	32	
...		

This atom is a list of ALAC frame sizes, each in bytes. For example, our 646 frame file would have 646 corresponding `stsz` entries.

the stco Atom

Field	Size	Value
atom length	32	offset × 4 + 16
atom type	32	'stco' (`0x7374636f`)
version	8	`0x00`
flags	24	`0x000000`
number of offsets	32	
frame offset	32	
...		

This atom is a list of absolute file offsets for each chunk, where each chunk is typically 5 ALAC frames large.

the udta Atom

Field	Size	Value
atom length	32	`meta` size + 8
atom type	32	'udta' (0x75647461)
`meta` atom	`meta` size	`meta` data

the meta Atom

Field	Size	Value
atom length	32	`hdlr` size + `ilst` size + `free` size + 12
atom type	32	'meta' (0x6d657461)
version	8	0x00
flags	24	0x000000
`hdlr` atom	`hdlr` size	`hdlr` data
`ilst` atom	`ilst` size	`ilst` data
`free` atom	`free` size	`free` data

the hdlr atom (revisited)

Field	Size	Value
atom length	32	34
atom type	32	'hdlr' (0x68646c72)
version	8	0x00
flags	24	0x000000
QuickTime type	32	0x00000000
QuickTime subtype	32	'mdir' (0x6d646972)
QuickTime manufacturer	32	'appl' (0x6170706c)
QuickTime component reserved flags	32	0x00000000
QuickTime component reserved flags mask	32	0x00000000
component name length	8	0x00
component name	0	

This atom is laid out identically to the ALAC file's primary `hdlr` atom (described on page 149). The only difference is the contents of its fields.

the ilst Atom

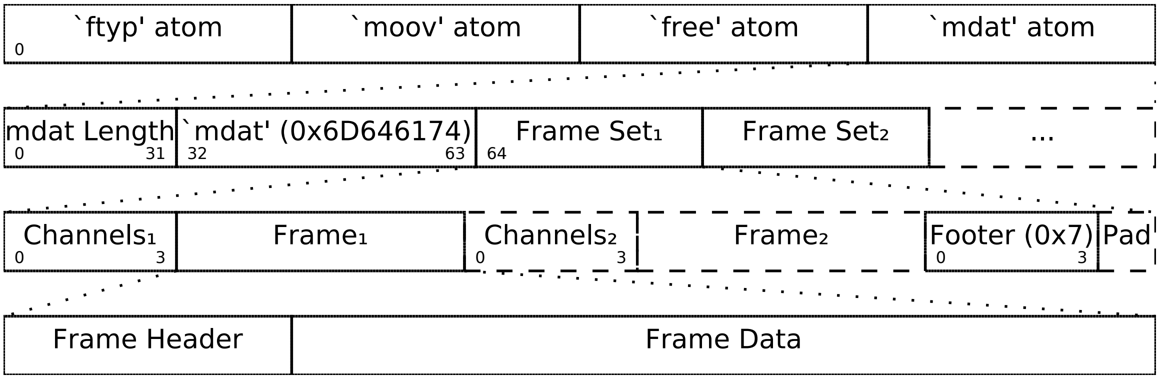
This atom is a collection of `data` sub-atoms and is described on page 133.

the free Atom

These atoms are simple collection of NULL bytes which can easily be resized to make room for other atoms without rewriting the entire file.

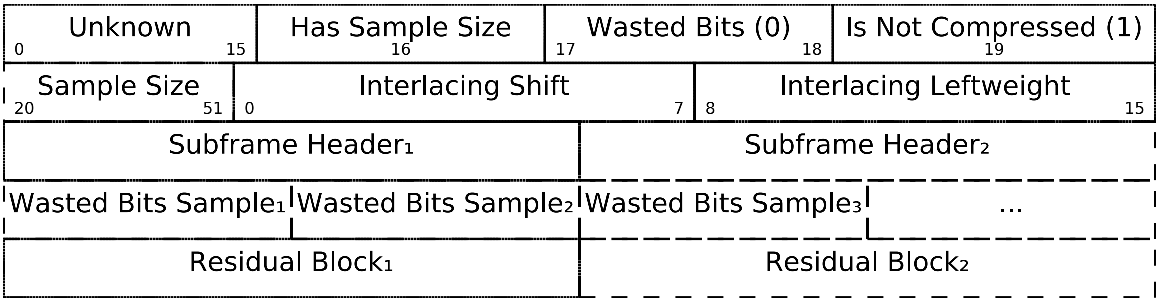
12.3.2 Compressed and Uncompressed Frames

Now that we've built a set of non-audio ALAC atoms, the next step is to break our audio into a set of PCM frames of a certain block size - typically 4096. When encoding more than 2 channels, these PCM frames then need to be broken apart into 1-2 channel sets such that each set becomes an ALAC frame and all the frames become an ALAC frameset.



We then have to decide whether to turn a set of 1-2 channel PCM frames into a compressed or uncompressed ALAC frame. This is done by first attempting a compressed frame while keeping track of its size. If that compressed frame's size is greater than what an uncompressed frame would be $((\text{block size} \times \text{channels} \times \text{bits per sample} \div 8) + 7)$, use an uncompressed frame instead.[3] For any audio data that isn't random noise, compressed frames will be the better choice.

As you'll recall, a compressed frame is laid out as follows:



'Has Sample Size' will be 0 so long as we have enough remaining samples to fill our consistent block size. But, at the end of the stream, this value will be 1 to cover any remaining samples whose size will be stored in the 'Sample Size' field. Next, we use a 'Wasted Bits' value of 1 if our stream's bits-per-sample is greater than 16.

In the event we have wasted bits, we simply chop off the bottom 8 bits of each sample and store them as a block of wasted bits samples (whose values are typically random noise,

[3] See page 136

which does not compress well) and then treat the stream as having 16 bits per sample throughout the remainder of compression.

12.3.3 Channel Correlation

For stereo streams, we must determine good 'Interlacing Shift' and 'Interlacing Leftweight' values to exploit similarities between the left and right channels.

In this case, we'll use an 'Interlacing Shift' value of 2 and try all all 'Interlacing Leftweight' values between 0 and 4 (inclusive), using the one that generates the smallest subframes.

Correlating our left and right channel samples using 'Interlacing Shift' and 'Interlacing Leftweight' is done using the inverse of decorrelation:

$$\text{Channel}_1 = \text{Right}_i + \left\lfloor \frac{(\text{Left}_i - \text{Right}_i) \times \text{Interlacing Leftweight}}{2^{\text{Interlacing Shift}}} \right\rfloor$$

$$\text{Channel}_2 = \text{Left}_i - \text{Right}_i$$

For example, given the left channels 18, 20, 26, 24, 24; the right channels of 2, 3, 0, -1. 0; an 'Interlacing Shift' value of 2 and an 'Interlacing Leftweight' value of 3, we calculate ALAC channels as follows:

Sample	Left_i	Right_i	Channel_1	Channel_2
0	18	2	$2 + \left\lfloor \frac{(18-2)\times 3}{2^2} \right\rfloor = 2 + 12 = \mathbf{14}$	$18 - 2 = \mathbf{16}$
1	20	3	$3 + \left\lfloor \frac{(20-3)\times 3}{2^2} \right\rfloor = 3 + 12 = \mathbf{15}$	$20 - 3 = \mathbf{17}$
2	26	0	$0 + \left\lfloor \frac{(26-0)\times 3}{2^2} \right\rfloor = 0 + 19 = \mathbf{19}$	$26 - 0 = \mathbf{26}$
3	24	-1	$-1 + \left\lfloor \frac{(24+1)\times 3}{2^2} \right\rfloor = -1 + 18 = \mathbf{17}$	$24 - -1 = \mathbf{25}$
4	24	0	$0 + \left\lfloor \frac{(24-0)\times 3}{2^2} \right\rfloor = 0 + 18 = \mathbf{18}$	$24 - 0 = \mathbf{24}$

12.3.4 Coefficient Calculation

Given a list of correlated samples, we need to generate one subframe header per channel by performing coefficient calculation to generate 'Prediction Quantitization', 'Coefficient Count' and a set of 'Coefficient' values. We'll set 'Prediction Type' to 0 and 'Rice Modifier' to 4.

Prediction Type 0 3	Prediction Quantitization 4 7	
Rice Modifier 8 10	Coefficient Count 11 15	
Coefficient1 16 31	Coefficient2 32 47	...

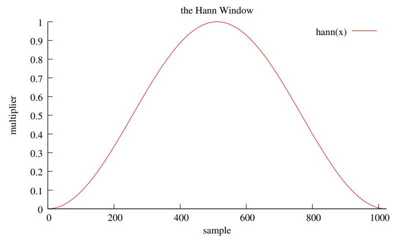
Windowing

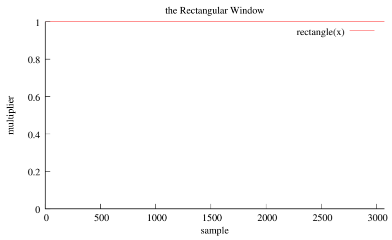
The first step in ALAC subframe encoding is ‘windowing’ the input signal. Put simply, this is a process of multiplying each input sample by an equivalent value from the window, which are floats from 0.0 to 1.0. In this case, the default is a Tukey window with a ratio of 0.5. A Tukey window is a combination of the Hann and Rectangular windows. The ratio of 0.5 means there’s 0.5 samples in the Hann window per sample in the Rectangular window.

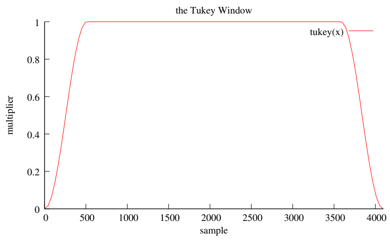
$$\text{hann}(n) = \frac{1}{2}\left(1 - \cos\left(\frac{2\pi n}{\text{sample count} - 1}\right)\right) \tag{12.6}$$

$$\text{rectangle}(n) = 1.0 \tag{12.7}$$

The Tukey window is defined by taking a Hann window, splitting it at the halfway point, and inserting a Rectangular window between the two.





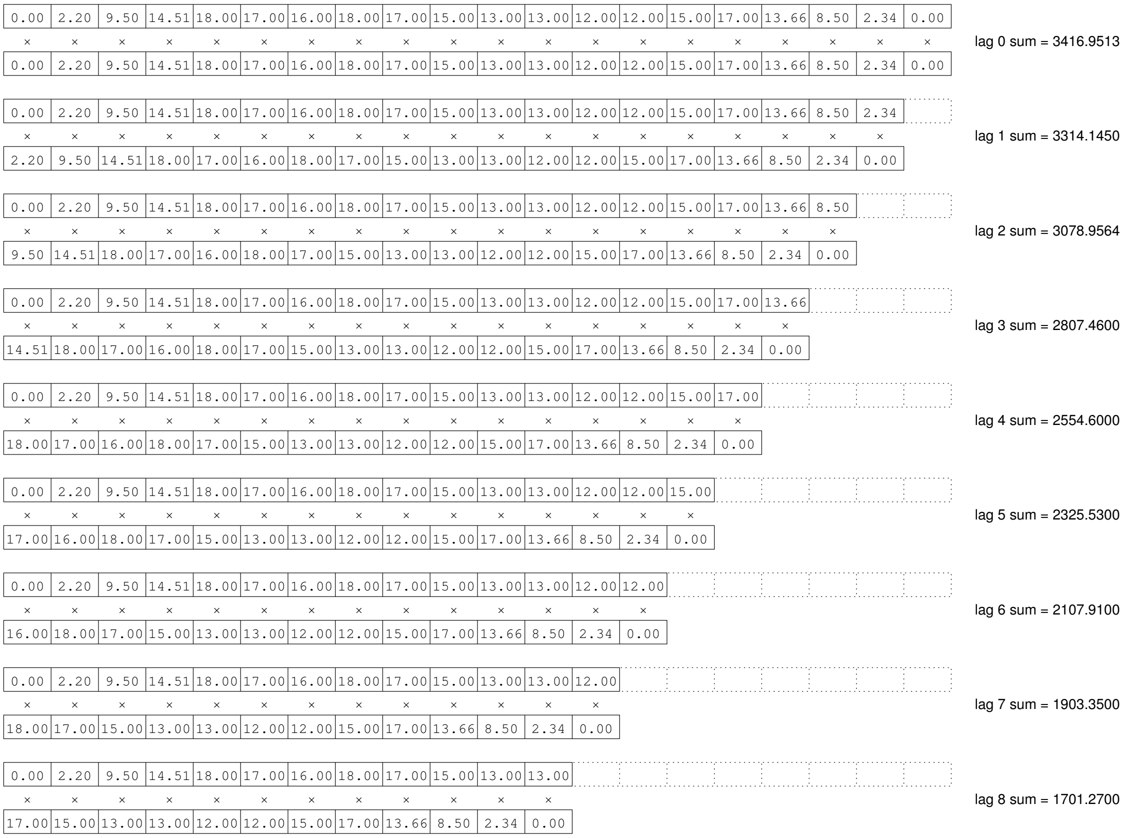


Let’s run through a short example with 20 samples:

index	input sample		Tukey window		windowed signal
0	14	×	0.0000	=	0.00
1	15	×	0.1464	=	2.20
2	19	×	0.5000	=	9.50
3	17	×	0.8536	=	14.51
4	18	×	1.0000	=	18.00
5	17	×	1.0000	=	17.00
6	16	×	1.0000	=	16.00
7	18	×	1.0000	=	18.00
8	17	×	1.0000	=	17.00
9	15	×	1.0000	=	15.00
10	13	×	1.0000	=	13.00
11	13	×	1.0000	=	13.00
12	12	×	1.0000	=	12.00
13	12	×	1.0000	=	12.00
14	15	×	1.0000	=	15.00
15	17	×	1.0000	=	17.00
16	16	×	0.8536	=	13.66
17	17	×	0.5000	=	8.50
18	16	×	0.1464	=	2.34
19	13	×	0.0000	=	0.00

Computing Autocorrelation

Once our input samples have been converted to a windowed signal, we then compute the autocorrelation values from that signal. Each autocorrelation value is determined by multiplying the signal's samples by the samples of a lagged version of that same signal, and then taking the sum. The lagged signal is simply the original signal with 'lag' number of samples removed from the beginning.



0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
× × × × × × × × × × × × × × × × × × × ×
0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 0 sum = 3416.9513

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34
× × × × × × × × × × × × × × × × × × ×
2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 1 sum = 3314.1450

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50
× × × × × × × × × × × × × × × × × ×
9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 2 sum = 3078.9564

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66
× × × × × × × × × × × × × × × × ×
14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 3 sum = 2807.4600

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00
× × × × × × × × × × × × × × × ×
18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 4 sum = 2554.6000

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00
× × × × × × × × × × × × × × ×
17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 5 sum = 2325.5300

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00
× × × × × × × × × × × × × ×
16.00 18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 6 sum = 2107.9100

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00 12.00
× × × × × × × × × × × × ×
18.00 17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 7 sum = 1903.3500

0.00 2.20 9.50 14.51 18.00 17.00 16.00 18.00 17.00 15.00 13.00 13.00
× × × × × × × × × × × ×
17.00 15.00 13.00 13.00 12.00 12.00 15.00 17.00 13.66 8.50 2.34 0.00
lag 8 sum = 1701.2700

In this example are autocorrelation values are: 3416.9513, 3314.1450, 3078.9564, 2807.4600, 2554.6000, 2325.5300, 2107.9100, 1903.3500 and 1701.2700.

LP Coefficient Calculation

Calculating the LP coefficients uses the Levinson-Durbin recursive method.[4] Our inputs are M, the maximum coefficient count, and r autocorrelation values, from $r(0)$ to $r(M-1)$. Our outputs are a, a list of LP coefficient lists from a_{11} to $a_{(M-1)(M-1)}$, and E, a list of error values from E_0 to $E_{(M-1)}$. q_m and κ_m are temporary values.
Initial values:

$$\begin{aligned} E_0 &= r(0) \\ a_{11} = \kappa_1 &= \frac{r(1)}{E_0} \\ E_1 &= E_0(1 - {\kappa_1}^2) \end{aligned}$$

With $m \geq 2$, the following recursive algorithm is performed:

Step 1. $$q_m = r(m) - \sum_{i=1}^{m-1} a_{i(m-1)} r(m-i)$$

Step 2. $$\kappa_m = \frac{q_m}{E_{(m-1)}}$$

Step 3. $a_{mm} = \kappa_m$

Step 4. $a_{im} = a_{i(m-1)} - \kappa_m a_{(m-i)(m-1)}$ for $i = 1$, $i = 2$,...,$i = m - 1$

Step 5. $E_m = E_{m-1}(1 - {\kappa_m}^2)$

Step 6. If $m < M$ then $m \leftarrow m + 1$ and goto step 1. If $m = M$ then stop.

Let's run through an example in which $M = 9$, $r(0) = 3417$, $r(1) = 3314$, $r(2) = 3079$, $r(3) = 2807$, $r(4) = 2555$, $r(5) = 2326$, $r(6) = 2108$, $r(7) = 1903$ and $r(8) = 1701$:

[4] This algorithm is taken from `http://www.engineer.tamuk.edu/SPark/chap7.pdf`

$$
\begin{aligned}
E_0 &= r(0) = 3417 \\
a_{11} &= \kappa_1 = \frac{r(1)}{E_0} = \frac{3314}{3417} = 0.97 \\
E_1 &= E_o(1 - \kappa_1{}^2) = 3417(1 - .97^2) = 201.95 \\
q_2 &= r(2) - \sum_{i=1}^{1} a_{i1} r(2-i) = 3079 - (0.97)(3314) = -135.58 \\
\kappa_2 &= \frac{q_2}{E_1} = \frac{-135.58}{201.95} = -0.671 \\
a_{22} &= \kappa_2 = -0.671 \\
a_{12} &= a_{11} - \kappa_2 a_{11} = 0.97 - (-0.671)(0.97) = 1.621 \\
E_2 &= E_1(1 - \kappa_2{}^2) = 201.95(1 - -0.671^2) = 111.024 \\
q_3 &= r(3) - \sum_{i=1}^{2} a_{i2} r(3-i) = 2807 - ((1.621)(3079) + (-0.671)(3314)) = 39.635 \\
\kappa_3 &= \frac{q_3}{E_2} = \frac{39.635}{111.024} = 0.357 \\
a_{33} &= \kappa_3 = 0.357 \\
a_{13} &= a_{12} - \kappa_3 a_{22} = 1.621 - (0.357)(-0.671) = 1.861 \\
a_{23} &= a_{22} - \kappa_3 a_{12} = -0.671 - (0.357)(1.621) = -1.25 \\
E_3 &= E_2(1 - \kappa_3{}^2) = 111.024(1 - 0.357^2) = 96.874 \\
q_4 &= r(4) - \sum_{i=1}^{3} a_{i3} r(4-i) = 2555 - ((1.861)(2807) + (-1.25)(3079) + (0.357)(3314)) = -3.175 \\
\kappa_4 &= \frac{q_4}{E_3} = \frac{-3.175}{96.874} = -0.033 \\
a_{44} &= \kappa_4 = -0.033 \\
a_{14} &= a_{13} - \kappa_4 a_{33} = 1.861 - (-0.033)(0.357) = 1.873 \\
a_{24} &= a_{23} - \kappa_4 a_{23} = -1.25 - (-0.033)(-1.25) = -1.291 \\
a_{34} &= a_{33} - \kappa_4 a_{33} = 0.357 - (-0.033)(1.861) = 0.418 \\
E_4 &= E_3(1 - \kappa_4{}^2) = 96.874(1 - -0.033^2) = 96.769
\end{aligned}
$$

Calculating E_5 through E_8 and a_{15} though a_{88} will be left as an exercise for the reader. Our final values are:

$a_{11} = 0.970$							
$a_{12} = 1.621$	$a_{22} = -0.67$						
$a_{13} = 1.861$	$a_{23} = -1.25$	$a_{33} = 0.357$					
$a_{14} = 1.873$	$a_{24} = -1.29$	$a_{34} = 0.418$	$a_{44} = -0.03$				
$a_{15} = 1.868$	$a_{25} = -1.28$	$a_{35} = 0.340$	$a_{45} = 0.122$	$a_{55} = -0.14$			
$a_{16} = 1.874$	$a_{26} = -1.28$	$a_{36} = 0.341$	$a_{46} = 0.119$	$a_{56} = -0.09$	$a_{66} = 0.00$		
$a_{17} = 1.875$	$a_{27} = -1.27$	$a_{37} = 0.332$	$a_{47} = 0.091$	$a_{57} = 0.011$	$a_{67} = -0.15$	$a_{77} = 0.078$	
$a_{18} = 1.892$	$a_{28} = -1.30$	$a_{38} = 0.338$	$a_{48} = 0.112$	$a_{58} = 0.082$	$a_{68} = -0.42$	$a_{78} = 0.479$	$a_{88} = -0.214$
$E_1 = 202.0$	$E_2 = 111.0$	$E_3 = 96.9$	$E_4 = 96.8$	$E_5 = 96.1$	$E_6 = 96.1$	$E_7 = 95.6$	$E_8 = 91.2$

These values have been rounded to the nearest significant digit and will not be an exact match to those generated by a computer.

Best Order Estimation

At this point, we have an array of prospective LP coefficient lists, a list of error values and must decide which LPC order to use. Making an estimation requires the total number of samples in the subframe, the number of overhead bits per order (by default, this is the number of bits per sample in the subframe, plus 5), and an error scale constant in addition to the LPC error values:

$$\text{Error Scale} = \frac{\ln(2)^2}{2 \times \text{Total Samples}} \tag{12.8}$$

Once the error scale has been calculated, one can generate a 'Bits per Residual' estimation function which, given an 'LPC Error' value, returns what its name implies:

$$\text{Bits per Residual(LPC Error)} = \frac{\ln(\text{Error Scale} \times \text{LPC Error})}{2 \times \ln(2)} \tag{12.9}$$

With this function, we can estimate how many bits the entire ALAC subframe will take for each 'LPC Error' value and its associated 'Order':

$$\text{Total Bits(LPC Error, Order)} = \text{Bits per Residual(LPC Error)} \times (\text{Total Samples} - \text{Order}) + (\text{Order} \times \text{Overhead bits})$$

Continuing with our example, we have 20 samples which gives us an error scale of: $\frac{\ln(2)^2}{2 \times 20} = \frac{.6931^2}{40} = .01201$

Now, we'll estimate the bits used by order 4 and order 8, which use LPC Error values 96.8 and 91.2, respectively:

At LPC order 4, our bits per residual are:

$$\frac{\ln(.01201 \times 96.8)}{2 \times \ln(2)} = \frac{\ln(1.163)}{1.386} = 0.1089$$

And our total bits are:

$$(0.1089 \times (20 - 1)) + (1 \times (16 + 5)) = 2.069 + 21 = 23.069$$

At LPC order 8, our bits per residual are:

$$\frac{\ln(.01201 \times 91.2)}{2 \times \ln(2)} = \frac{\ln(1.095)}{1.386} = 0.065$$

And our total bits are:

$$(0.065 \times (20 - 2)) + (2 \times (16 + 5)) = 1.17 + 42 = 43.17$$

Therefore, since the total bits for order 4 are the smallest, the best order for this group of samples is 4.

Quantizing Coefficients

Quantizing coefficients is a process of taking a list of LP Coefficients along with a QLP Coefficients Precision value and returning a list of Coefficients and a Prediction Quantitization value. The first step is determining the upper and lower limits of the Coefficients, which well set to the upper and lower bounds that iTunes supports:

$$\text{Coefficient Maximum} = 2^{12-1} - 1 = 2047 \tag{12.10}$$
$$\text{Coefficient Minimum} = -2^{12-1} = -2048 \tag{12.11}$$

Prediction Quantitization is always 9 in iTunes, so we'll use that value also:

$$\text{Prediction Quantitization} = 9 \tag{12.12}$$

We determine the Coefficient values themselves via a small recursive routine:

$$X(i) = E(i-1) + (\text{LP Coefficient}_i \times 2^{quantitization}) \tag{12.13}$$
$$\text{Coefficient}_i = \text{round}(X(i)) \tag{12.14}$$
$$E(i) = X(i) - \text{Coefficient}_i \tag{12.15}$$

where $E(0) = 0$ and each Coefficient is adjusted prior to calculating the next $E(i)$ value such that:

$$\text{Coefficient Minimum} \leq \text{Coefficient}_i \leq \text{Coefficient Maximum} \tag{12.16}$$

So to finish our example in which we're quantizing the LP coefficients 1.873, -1.29, 0.418 and -0.03:

$$\begin{aligned}
X(1) &= E(0) + (1.873 \times 2^9) = 0 + 958.976 = \mathbf{958.976} \\
\text{QLP Coefficient}_1 &= \text{round}(958.976) = \mathbf{959} \\
E(1) &= X(1) - \text{QLP Coefficient}_1 = 958.976 - 959 = \mathbf{-0.024} \\
X(2) &= E(1) + (-1.29 \times 2^9) = -0.024 + -660.48 = \mathbf{-660.504} \\
\text{QLP Coefficient}_2 &= \text{round}(-660.504) = \mathbf{-661} \\
E(2) &= X(2) - \text{QLP Coefficient}_2 = -660.504 - -661 = \mathbf{0.496} \\
X(3) &= E(2) + (0.418 \times 2^9) = 0.496 + 214.015 = \mathbf{214.511} \\
\text{QLP Coefficient}_3 &= \text{round}(214.511) = \mathbf{215} \\
E(3) &= X(3) - \text{QLP Coefficient}_3 = 214.511 - 215 = \mathbf{-0.489} \\
X(4) &= E(3) + (-0.03 \times 2^9) = -0.489 + -15.36 = \mathbf{-15.849} \\
\text{QLP Coefficient}_4 &= \text{round}(-15.849) = \mathbf{-16}
\end{aligned}$$

Therefore, the 'Coefficient Count' is 4, the 'Coefficient' values are 959, -661, 215, -16, and the 'Prediction Quantitization' value is 9.

12.3.5 Subframe Calculation

Given our list of correlated samples; along with a list of coefficients, a ‘Coefficient Count’ and a ‘Prediction Quantitization’ value, we can now generate a list of signed residuals values for a given channel.

The first residual is always the first input sample:

$$\text{Residual}_0 = \text{Sample}_0$$

Then, for the next ‘Coefficient Count’ number of samples:

$$\text{Residual}_i = \text{Sample}_i - \text{Sample}_{i-1}$$

For example, given that we have a ‘Coefficient Count’ of 4 and our first five samples are 16, 17, 26, 25 and 24; our first five residual values are:

$$\begin{aligned}
\text{Residual}_0 &= \mathbf{16} \\
\text{Residual}_1 &= 17 - 16 = \mathbf{1} \\
\text{Residual}_2 &= 26 - 17 = \mathbf{9} \\
\text{Residual}_3 &= 25 - 26 = \mathbf{\text{-}1} \\
\text{Residual}_4 &= 24 - 25 = \mathbf{\text{-}1}
\end{aligned}$$

These are our “starting point” residuals upon which the remainder of the residuals will be calculated from.

Subsequent residuals are calculated in the following way:

$$\begin{aligned}
\text{LPC Sum}_i &= \sum_{j=0}^{coeffs-1} \text{Coefficient}_j \times (\text{Sample}_{i-j-1} - \text{Sample}_{i-coeffs-1}) \\
\text{Residual}_i &= \text{Sample}_i - \left(\left\lfloor \frac{\text{LPC Sum}_i + 2^{\text{Predictor Quantitization - 1}}}{2^{\text{Predictor Quantitization}}} \right\rfloor + \text{Sample}_{i-coeffs-1} \right)
\end{aligned}$$

For example, given $\text{Sample}_5 = 23$, ‘Predictor Quantitization’ = 9 and the coefficients 1122, -766, 107 and 122:

$$\begin{aligned}
\text{LPC Sum}_5 &= (1122 \times (24 - 16)) + (-766 \times (25 - 16)) + (107 \times (26 - 16)) + (122 \times (17 - 16)) \\
&= (1122 \times 8) + (-766 \times 9) + (107 \times 10) + (122 \times 1) \\
&= 8976 + -6894 + 1070 + 122 = \mathbf{3274} \\
\text{Residual}_5 &= 23 - \left(\left\lfloor \frac{\mathbf{3274} + 2^8}{2^9} \right\rfloor + 16 \right) = 23 - \left(\left\lfloor \frac{3530}{512} \right\rfloor + 16 \right) = 23 - (6 + 16) = \mathbf{1}
\end{aligned}$$

But before calculation Residual_6, we need to adjust our coefficient list in the same way as residual decoding.

As described on page 142, we run through the following process:

Step 1.	original sign	$=$	$\text{sign}(residual)$
Step 2.	j	$=$	0
Step 3.	v	$=$	$\text{Sample}_{i-\text{count}-1} - \text{Sample}_{i-\text{count}+j}$
Step 4.	$sign$	$=$	$\begin{cases} \text{sign}(v) & \text{if original sign} > 0 \\ -\text{sign}(v) & \text{if original sign} \leq 0 \end{cases}$
Step 5.	$\text{Coefficient}_{\text{count}-j-1}$	$=$	$\text{Coefficient}_{\text{count}-j-1} - sign$
Step 6.	$residual$	$=$	$residual - \left(\left\lfloor \frac{v \times sign}{2^{\text{Predictor Quantitization}}} \right\rfloor \times (j+1)\right)$
Step 7.	j	$=$	$j+1$
Step 8.	if ($j < count$) and ($\text{sign}(residual) = \text{original sign}$), goto step 3		

To continue our example:

$$
\begin{aligned}
\text{original sign} &= \text{sign}(1) = \mathbf{1} \\
j &= 0 \\
v &= \text{Sample}_0 - \text{Sample}_1 = 16 - 17 = \textbf{-1} \\
sign &= \text{sign}(-1) = \textbf{-1} \\
\text{Coefficient}_3 &= \text{Coefficient}_3 - -1 = 122 + 1 = \mathbf{123} \\
residual &= residual - \left(\left\lfloor \frac{-1 \times -1}{2^9} \right\rfloor \times (0+1)\right) = 1 - (0 \times 1) = \mathbf{1} \\
j &= 1 \\
v &= \text{Sample}_0 - \text{Sample}_2 = 16 - 26 = \textbf{-10} \\
sign &= \text{sign}(-10) = \textbf{-1} \\
\text{Coefficient}_2 &= \text{Coefficient}_2 - -1 = 107 + 1 = \mathbf{108} \\
residual &= residual - \left(\left\lfloor \frac{-10 \times -1}{2^9} \right\rfloor \times (1+1)\right) = 1 - (0 \times 2) = \mathbf{1} \\
j &= 2 \\
v &= \text{Sample}_0 - \text{Sample}_3 = 16 - 25 = \textbf{-9} \\
sign &= \text{sign}(-9) = \textbf{-1} \\
\text{Coefficient}_1 &= \text{Coefficient}_1 - -1 = -766 + 1 = \textbf{-765} \\
residual &= residual - \left(\left\lfloor \frac{-9 \times -1}{2^9} \right\rfloor \times (2+1)\right) = 1 - (0 \times 3) = \mathbf{1} \\
j &= 3 \\
v &= \text{Sample}_0 - \text{Sample}_4 = 16 - 25 = \textbf{-8} \\
sign &= \text{sign}(-8) = \textbf{-1} \\
\text{Coefficient}_0 &= \text{Coefficient}_0 - -1 = 1122 + 1 = \mathbf{1123} \\
residual &= residual - \left(\left\lfloor \frac{-8 \times -1}{2^9} \right\rfloor \times (3+1)\right) = 1 - (0 \times 4) = \mathbf{1} \\
j &= 4 \text{ and stop, since 4 equals our coefficient count}
\end{aligned}
$$

Which, you'll notice, is identical to the procedure used during residual decoding.

Given that Sample_6 is 20, the calculation of Residual_6 is as follows:

$$\begin{aligned}
\text{LPC Sum}_6 &= (1123 \times (23 - 17)) + (-765 \times (24 - 17)) + (108 \times (25 - 17)) + (123 \times (26 - 17)) \\
&= (1123 \times 6) + (-765 \times 7) + (108 \times 8) + (123 \times 9) \\
&= 6738 + -5355 + 864 + 1107 = \mathbf{3354} \\
\text{Residual}_6 &= 20 - \left(\left\lfloor \frac{\mathbf{3354} + 2^8}{2^9} \right\rfloor + 17\right) = 20 - \left(\left\lfloor \frac{3610}{512} \right\rfloor + 17\right) = 20 - (7 + 17) = \mathbf{\text{-}4} \\
\text{original sign} &= \text{sign}(-4) = -1 \\
j &= 0 \\
v &= \text{Sample}_1 - \text{Sample}_2 = 17 - 26 = \mathbf{\text{-}9} \\
\textit{sign} &= -\text{sign}(-9) = \mathbf{1} \\
\text{Coefficient}_3 &= \text{Coefficient}_3 - 1 = 123 - 1 = \mathbf{122} \\
\textit{residual} &= \textit{residual} - \left(\left\lfloor \frac{-9 \times 1}{2^9} \right\rfloor \times (0 + 1)\right) = -4 - (-1 \times 1) = \mathbf{\text{-}3} \\
j &= 1 \\
v &= \text{Sample}_1 - \text{Sample}_3 = 17 - 25 = \mathbf{\text{-}8} \\
\textit{sign} &= -\text{sign}(-8) = \mathbf{1} \\
\text{Coefficient}_2 &= \text{Coefficient}_2 - 1 = 108 - 1 = \mathbf{107} \\
\textit{residual} &= \textit{residual} - \left(\left\lfloor \frac{-8 \times 1}{2^9} \right\rfloor \times (1 + 1)\right) = -3 - (-1 \times 2) = \mathbf{\text{-}1} \\
j &= 2 \\
v &= \text{Sample}_1 - \text{Sample}_4 = 17 - 24 = \mathbf{\text{-}7} \\
\textit{sign} &= -\text{sign}(-7) = \mathbf{1} \\
\text{Coefficient}_1 &= \text{Coefficient}_1 - 1 = -765 - 1 = \mathbf{\text{-}766} \\
\textit{residual} &= \textit{residual} - \left(\left\lfloor \frac{-7 \times 1}{2^9} \right\rfloor \times (2 + 1)\right) = -1 - (-1 \times 3) = \mathbf{2} \\
& \text{and stop, since sign(2)} \neq \text{sign}(-4) \text{ (our `original sign' value)}
\end{aligned}$$

So, the coefficients for Residual_7 are 1123, -766, 107 and 122.

12.3.6 Residual Encoding

The final step of frame encoding is writing our list of residual values. As with decoding, this also requires knowing several additional values whose defaults are as follows:

value	default
Initial History	10
History Multiplier	40
Maximum K	14
Bits per Sample	Stream's BPS − (Wasted Bits × 8) + Channels − 1

For each residual block, 'History' starts with the value of 'Initial History' and will change during residual encoding. We use it to calculate κ using the following formula:

$$\kappa = \left\lfloor \log_2 \left(\frac{\text{history}}{2^9} + 3 \right) \right\rfloor \tag{12.17}$$

Note that if κ exceeds the 'Maximum K' value from the `alac` atom, 'Maximum K' is used instead.

The κ and 'Bits per Sample' values are used to write a single unsigned residual value r in the following way:

First, we convert our signed residual to an unsigned value:

$$\text{unsigned} = \begin{cases} \text{signed} \times 2 & \text{if signed} \geq 0 \\ (-\text{signed} \times 2) - 1 & \text{if signed} < 0 \end{cases}$$

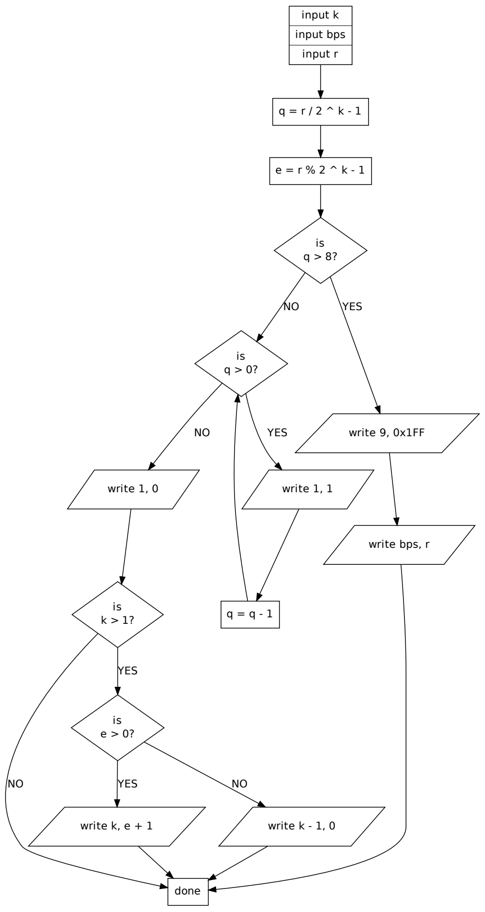
We then divide our unsigned residual into quotient and remainder values:

$$q = \lfloor r \div (2^k - 1) \rfloor$$
$$e = r \mod 2^k - 1$$

If q is greater than 8, we write a 9 bit value `0x1FF` (i.e. the bits '`1 1 1 1 1 1 1 1 1`') before writing the value r verbatim using 'Bits per Sample' number of bits.

If q is greater than 0, we write its value in unary. Otherwise, we write a single `0` bit.

Finally, if e is greater than 0, we write the value $e + 1$ using κ of bits. Otherwise, we write the value 0 using $\kappa - 1$ number of bits. Assuming κ is greater than 1, of course.



As with decoding, we then use our unsigned value to update 'history' before encoding the next residual:

$$\text{history} = \begin{cases} \text{history} + (\text{unsigned} \times \text{history multiplier}) - \lfloor \frac{\text{history} \times \text{history multiplier}}{2^9} \rfloor & \text{if unsigned} \leq 65535 \\ 65535 & \text{if unsigned} > 65535 \end{cases}$$

Should history ever fall below 128, we generate a special residual value. That residual's 'Bits per Sample' value is 16, its κ value is:

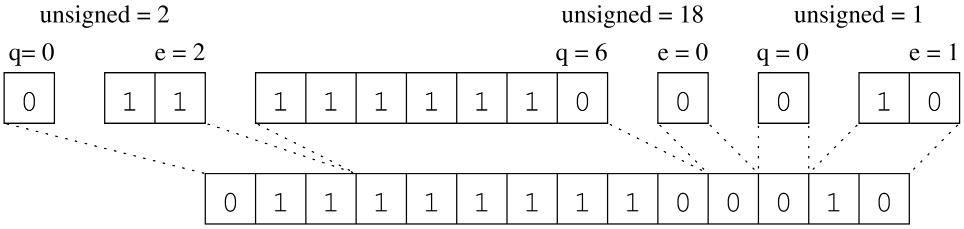
$$\kappa_{\text{blocksize}} = 7 - \log_2(\text{history}) + \frac{\text{history} + 16}{64} \qquad (12.18)$$

and its value is how many consecutive '0' residuals follow - which may be 0 if the next residual is not '0'. Either way, if 'block size' $\leq$ 65535, we subtract 1 to the next unsigned value of the next residual in the block (if any). Finally, 'history' is automatically set to 0.

Residual Encoding Example

In this example, we'll encode a group of residuals in which our 'Initial History' is 1290, our 'History Multiplier' is 40 and our signed residual values are 1, 9 and -1:

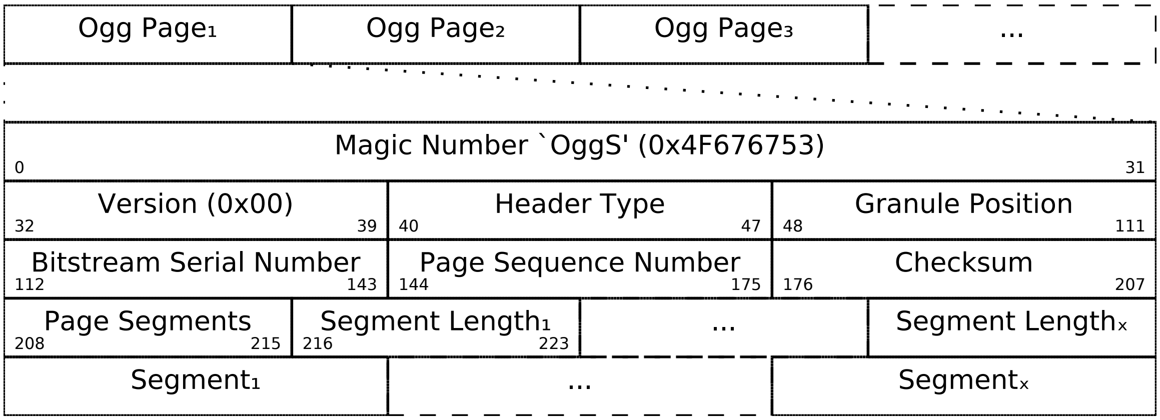
- Residual 1
  - $\kappa = \lfloor \log_2((1290 \div 2^9) + 3) \rfloor = \lfloor \log_2(5) \rfloor = 2$
  - $\text{unsigned}_1 = 2 \times 1 = \mathbf{2}$
  - $\text{history} = 1290 + (2 \times 40) - \lfloor (1290 \times 40) \div 2^9 \rfloor = 1370 - 100 = 1270$
- Residual 2
  - $\kappa = \lfloor \log_2((1270 \div 2^9) + 3) \rfloor = \lfloor \log_2(5) \rfloor = 2$
  - $\text{unsigned}_2 = 2 \times 9 = \mathbf{18}$
  - $\text{history} = 1270 + (18 \times 40) - \lfloor (1270 \times 40) \div 2^9 \rfloor = 1990 - 99 = 1891$
- Residual 3
  - $\kappa = \lfloor \log_2((1891 \div 2^9) + 3) \rfloor = \lfloor \log_2(6) \rfloor = 2$
  - $\text{unsigned}_3 = (1 \times 2) - 1 = \mathbf{1}$
  - $\text{history} = 1891 + (1 \times 40) - \lfloor (1891 \times 40) \div 2^9 \rfloor = 1931 - 147 = 1784$



13 Ogg Vorbis

Ogg Vorbis is Vorbis audio in an Ogg container. Ogg containers are a series of Ogg pages, each containing one or more segments of data. All of the fields within Ogg Vorbis are little-endian.

13.1 Ogg File Stream



bits	Header Type
001	Continuation
010	Beginning of Stream
100	End of Stream

‘Granule position’ is a time marker. In the case of Ogg Vorbis, it is the sample count.

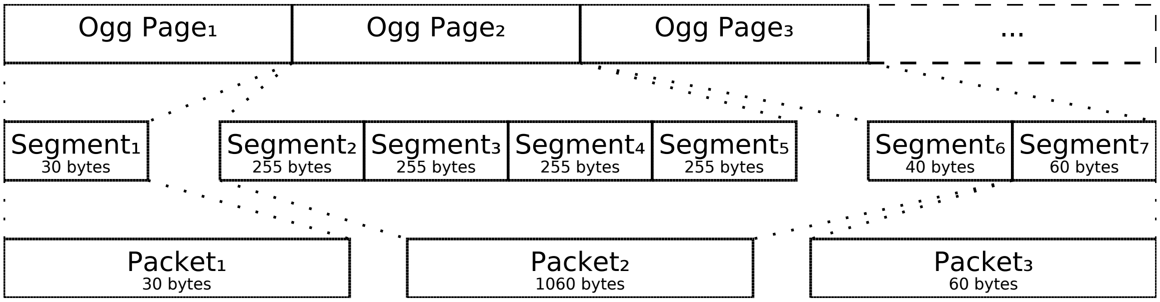
‘Bitstream Serial Number’ is an identifier for the given bitstream which is unique within the Ogg file. For instance, an Ogg file might contain both video and audio pages, interleaved. The Ogg pages for the audio will have a different serial number from those of the video so that the decoder knows where to send the data of each.

‘Page Sequence Number’ is an integer counter which starts from 0 and increments 1 for each Ogg page. Multiple bitstreams will have separate sequence numbers.

‘Checksum’ is a 32-bit checksum of the entire Ogg page.

The ‘Page Segments’ value indicates how many segments are in this Ogg page. Each segment will have an 8-bit length. If that length is 255, it indicates the next segment is part of the current one and should be concatenated with it when creating packets from the segments. In this way, packets larger than 255 bytes can be stored in an Ogg page. If the final segment in the Ogg page has a length of 255 bytes, the packet it is a part of continues into the next Ogg page.

13.1.1 Ogg Packets



This is an example Ogg stream to illustrate a few key points about the format. Note that Ogg pages may have one or more segments, and packets are composed of one of more segments, yet the boundaries between packets are segments that are less than 255 bytes long. Which segment belongs to which Ogg page is not important for building packets.

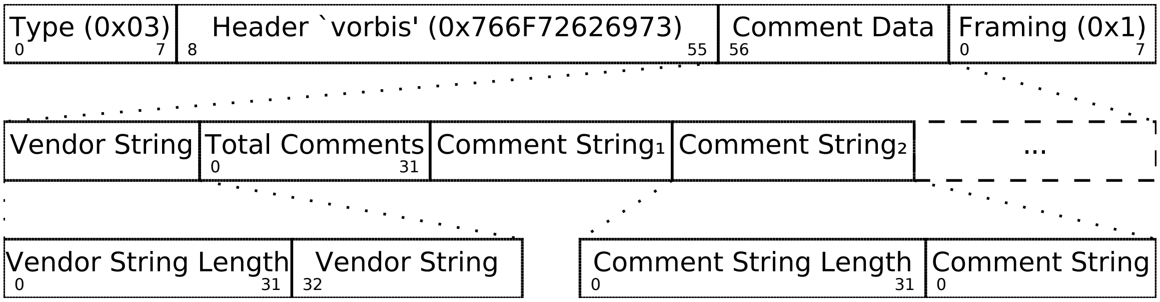
13.2 the Identification Packet

The first packet within a Vorbis stream is the Identification packet. This contains the sample rate and number of channels. Vorbis does not have a bits-per-sample field, as samples are stored internally as floating point values and are converted into a certain number of bits in the decoding process. To find the total samples, use the 'Granule Position' value in the stream's final Ogg page.

Type (0x01)	0	7
Header `vorbis' (0x766F72626973)	8	55
Vorbis version (0x00000000)	56	87
Channels	88	95
Sample Rate	96	127
Maximum Bitrate	128	159
Nominal Bitrate	160	191
Minimum Bitrate	192	223
Blocksize₁	224	227
Blocksize₂	228	231
Framing flag (0x01)	232	239

13.3 the Comment Packet

The second packet within a Vorbis stream is the Comment packet.



The length fields are all little-endian. The ‘Vendor String’ and ‘Comment Strings’ are all UTF-8 encoded. Keys are not case-sensitive and may occur multiple times, indicating multiple values for the same field. For instance, a track with multiple artists may have more than one `ARTIST`.

ALBUM	album name
ARTIST	artist name, band name, composer, author, etc.
CATALOGNUMBER*	CD spine number
COMPOSER*	the work’s author
CONDUCTOR*	performing ensemble’s leader
COPYRIGHT	copyright attribution
DATE	recording date
DESCRIPTION	a short description
DISCNUMBER*	disc number for multi-volume work
ENGINEER*	the recording masterer
ENSEMBLE*	performing group
GENRE	a short music genre label
GUEST ARTIST*	collaborating artist
ISRC	ISRC number for the track
LICENSE	license information
LOCATION	recording location
OPUS*	number of the work
ORGANIZATION	record label
PART*	track’s movement title
PERFORMER	performer name, orchestra, actor, etc.
PRODUCER*	person responsible for the project
PRODUCTNUMBER*	UPC, EAN, or JAN code
PUBLISHER*	album’s publisher
RELEASE DATE*	date the album was published
REMIXER*	person who created the remix
SOURCE ARTIST*	artist of the work being performed
SOURCE MEDIUM*	CD, radio, cassette, vinyl LP, etc.
SOURCE WORK*	a soundtrack’s original work
SPARS*	DDD, ADD, AAD, etc.
SUBTITLE*	for multiple track names in a single file
TITLE	track name
TRACKNUMBER	track number
VERSION	track version

Fields marked with * are proposed extension fields and not part of the official Vorbis comment specification.

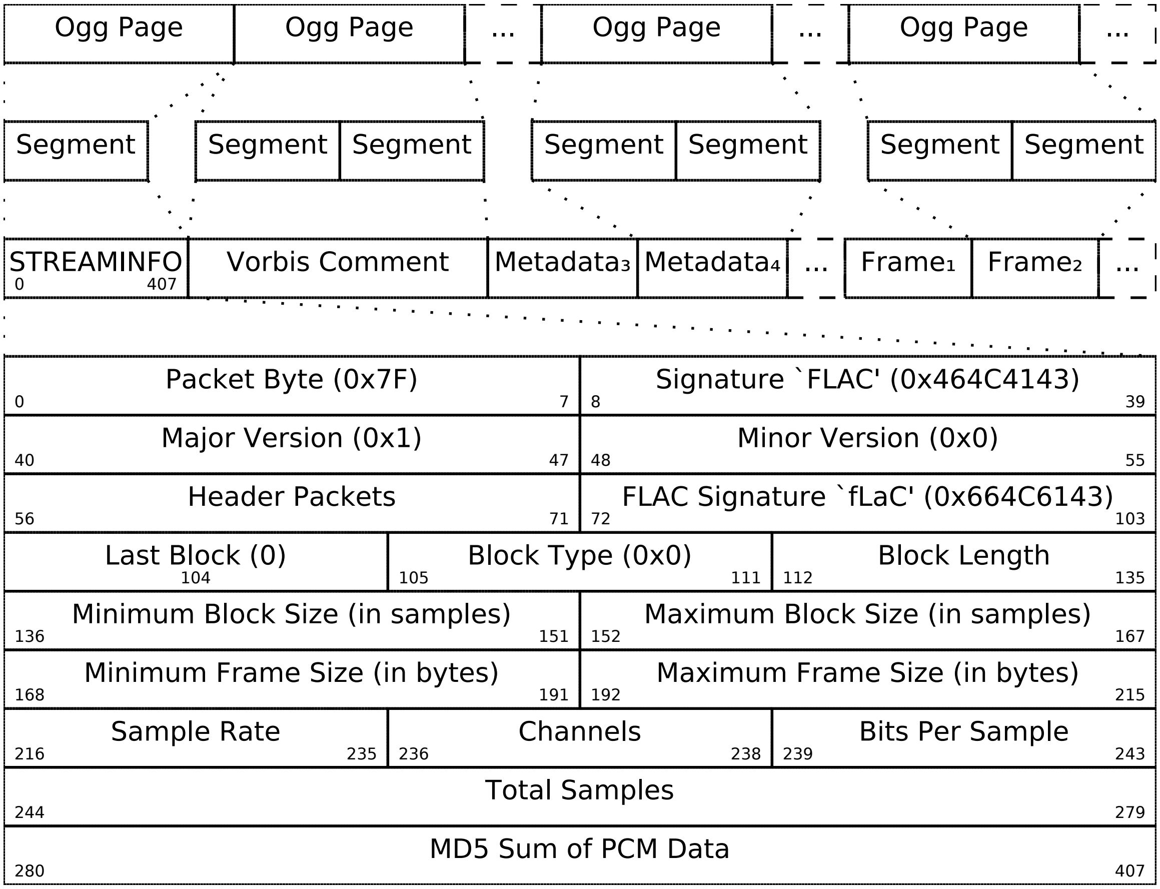
13.4 Channel Assignment

channel count	channel 1	channel 2	channel 3	channel 4	channel 5	channel 6	channel 7	channel 8
1	front center							
2	front left	front right						
3	front left	front center	front right					
4	front left	front right	back left	back right				
5	front left	front center	front right	back left	back right			
6	front left	front center	front right	back left	back right	LFE		
7	front left	front center	front right	side left	side right	back center	LFE	
8	front left	front center	front right	side left	side right	back left	back right	LFE
8+	defined by application							

14 Ogg FLAC

Ogg FLAC is a FLAC audio stream in an Ogg container.

14.1 the Ogg FLAC File Stream



Packet Byte (0x7F)	0	7
Signature `FLAC' (0x464C4143)	8	39
Major Version (0x1)	40	47
Minor Version (0x0)	48	55
Header Packets	56	71
FLAC Signature `fLaC' (0x664C6143)	72	103
Last Block (0)	104	
Block Type (0x0)	105	111
Block Length	112	135
Minimum Block Size (in samples)	136	151
Maximum Block Size (in samples)	152	167
Minimum Frame Size (in bytes)	168	191
Maximum Frame Size (in bytes)	192	215
Sample Rate	216	235
Channels	236	238
Bits Per Sample	239	243
Total Samples	244	279
MD5 Sum of PCM Data	280	407

Subsequent FLAC metadata blocks are stored 1 per packet. Each contains the 32-bit FLAC metadata block header in addition to the metadata itself. The `VORBIS_COMMENT` metadata block is required to immediately follow the `STREAMINFO` block, but all others may appear in any order.

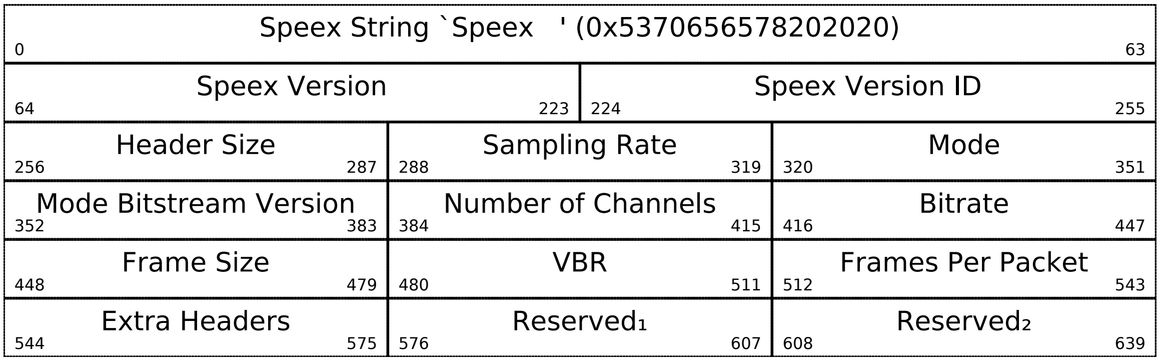
15 Ogg Speex

Ogg Speex is Speex audio in an Ogg container. Speex is a lossy audio codec optimized for speech. All of the fields within Ogg Speex are little-endian.

How Ogg containers break up data packets into segments and pages has already been explained in the Ogg Vorbis section on page 169. Therefore, I shall move directly to the Ogg Speex packets themselves.

15.1 the Header Packet

The first packet within a Speex stream is the Header packet. It contains the number of channels and sampling rate. Like Vorbis, the number of bits per sample is generated during decoding and the total number of samples is pulled from the 'Granule Position' field in the Ogg stream.



<table>
<tr><td colspan="6">Speex String `Speex   ' (0x5370656578202020)
0 63</td></tr>
<tr><td colspan="3">Speex Version
64 223</td><td colspan="3">Speex Version ID
224 255</td></tr>
<tr><td colspan="2">Header Size
256 287</td><td colspan="2">Sampling Rate
288 319</td><td colspan="2">Mode
320 351</td></tr>
<tr><td colspan="2">Mode Bitstream Version
352 383</td><td colspan="2">Number of Channels
384 415</td><td colspan="2">Bitrate
416 447</td></tr>
<tr><td colspan="2">Frame Size
448 479</td><td colspan="2">VBR
480 511</td><td colspan="2">Frames Per Packet
512 543</td></tr>
<tr><td colspan="2">Extra Headers
544 575</td><td colspan="2">Reserved$_1$
576 607</td><td colspan="2">Reserved$_2$
608 639</td></tr>
</table>

15.2 the Comment Packet

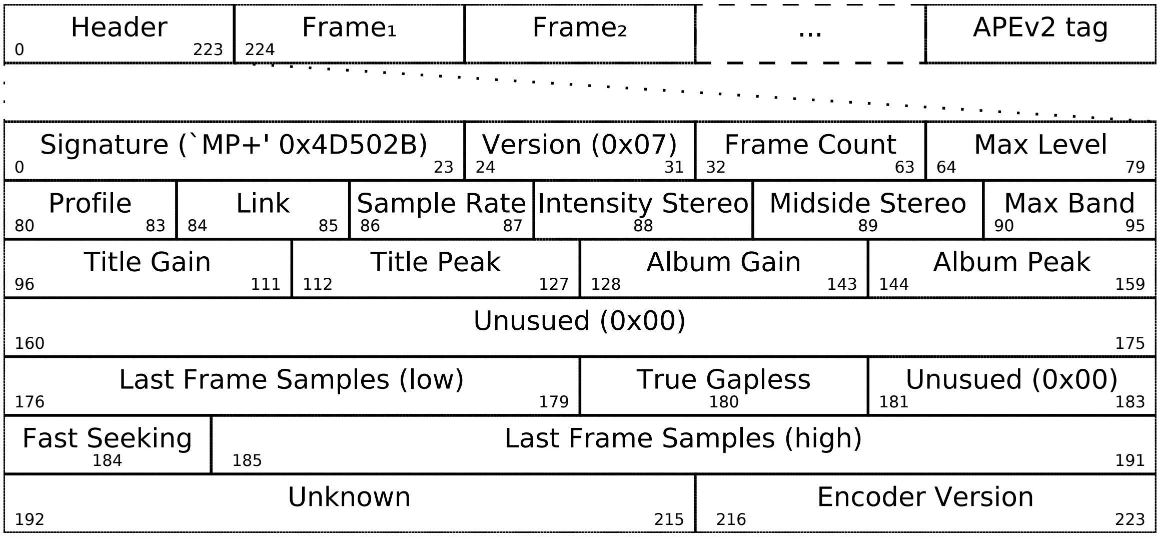
The second packet within a Speex stream is the Comment packet. This is identical to the comments used by Ogg Vorbis which is detailed on page 171.

16 Musepack

Musepack is a lossy audio format based on MP2 and designed for transparency. It comes in two varieties: SV7 and SV8 where 'SV' stands for Stream Version. These container versions differ so heavily that they must be considered separately from one another.

16.1 the SV7 File Stream

This is the earliest version of Musepack with wide support. All of its fields are little-endian. Each frame contains 1152 samples per channel. Therefore:



Header 0 223 | Frame₁ 224 | Frame₂ | ... | APEv2 tag

Signature (`MP+' 0x4D502B) 0 23 | Version (0x07) 24 31 | Frame Count 32 63 | Max Level 64 79

Profile 80 83 | Link 84 85 | Sample Rate 86 87 | Intensity Stereo 88 | Midside Stereo 89 | Max Band 90 95

Title Gain 96 111 | Title Peak 112 127 | Album Gain 128 143 | Album Peak 144 159

Unusued (0x00) 160 175

Last Frame Samples (low) 176 179 | True Gapless 180 | Unusued (0x00) 181 183

Fast Seeking 184 | Last Frame Samples (high) 185 191

Unknown 192 215 | Encoder Version 216 223

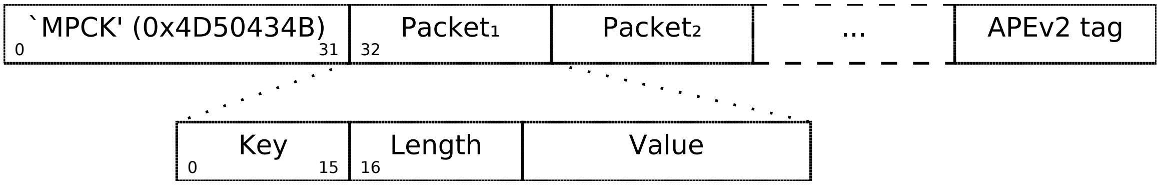
$$\text{Total Samples} = ((\text{Frame Count} - 1) \times 1152) + \text{Last Frame Samples} \tag{16.1}$$

Musepack files always have exactly 2 channels and its lossy samples are stored as floating point. Its sampling rate is one of four values:

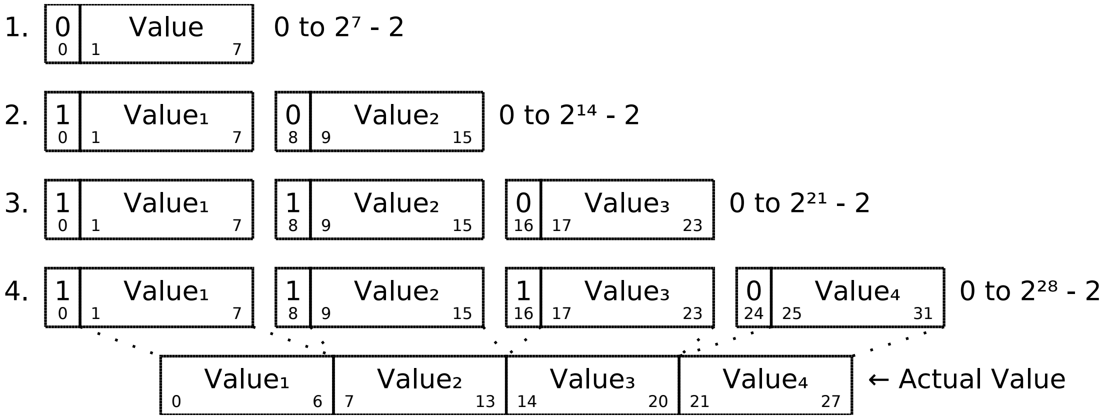
$\texttt{00} = 44100\text{Hz}$, $\texttt{01} = 48000\text{Hz}$, $\texttt{10} = 37800\text{Hz}$, $\texttt{11} = 32000\text{Hz}$.

16.2 the SV8 File Stream

This is the latest version of the Musepack stream. All of its fields are big-endian.

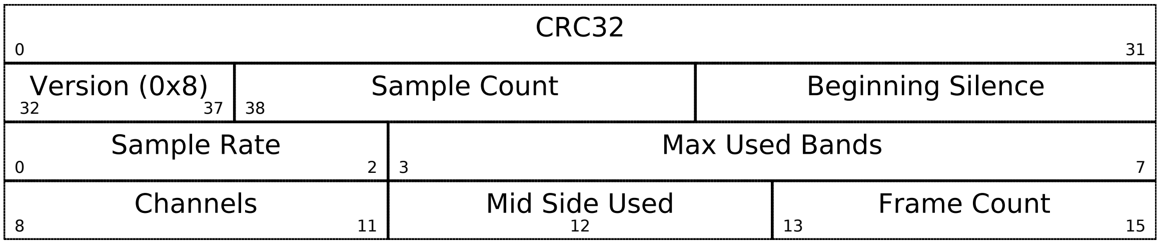


‘Key’ is a two character uppercase ASCII string (i.e. each digit must be between the characters 0x41 and 0x5A, inclusive). ‘Length’ is a variable length field indicating the size of the entire packet, including the header. This is a Nut-encoded field whose total size depends on whether the eighth bit of each byte is 0 or 1. The remaining seven bits of each byte combine to form the field’s value, which is big-endian.



16.2.1 the SH Packet

This is the Stream Header, which must be found before the first audio packet in the file.



‘CRC32’ is a checksum of everything in the header, not including the checksum itself. ‘Sample Count’ is the total number of samples, as a Nut-encoded value. ‘Beginning Silence’ is the number of silence samples at the start of the stream, also as a Nut-encoded value. ‘Channels’ is the total number of channels in the stream, minus 1. ‘Mid Side Used’ indicates the channels are stored using mid-side stereo. ‘Frame Count’ is used to calculate the total number of frames per audio packet:

$$\text{Number of Frames} = 4^{\text{Frame Count}} \tag{16.2}$$

‘Sample Rate’ is one of four values:
`000` = 44100Hz, `001` = 48000Hz, `010` = 37800Hz, `011` = 32000Hz .

16.2.2 the SE Packet

This is an empty packet that denotes the end of the Musepack stream. A decoder should ignore everything after this packet, which allows for metadata tags such as APEv2 to be placed at the end of the file.

16.2.3 the RG Packet

This is ReplayGain information about the file.

Version (0x1) 0 7	
Title Gain 8 23	Title Peak 24 39
Album Gain 40 55	Album Peak 56 71

16.2.4 the EI Packet

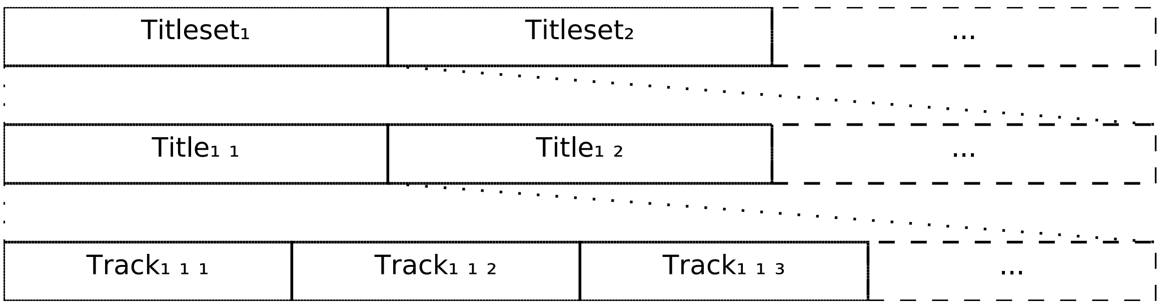
This is information about the Musepack encoder.

Profile 0 6	PNS 7	Major Version 8 15
Minor Version 16 23		Build 24 31

17 DVD-Audio

DVD-Audio is a format for delivering hi-fidelity, multichannel audio on DVD media. A DVD-Audio's `AUDIO_TS` directory contains the relevent data needed for decoding, spread into a lot of files whose names are more than a little cryptic at first glance.

Unlike CD audio, which is simply a set of 1 to 99 identically-formatted audio tracks (in terms of channel count, sample rate and bits per sample), a DVD-Audio disc contains one or more titlesets. Each titleset contains one or more titles, and each title contains one or more tracks.



Typically, a DVD-Audio disc will contain two titlesets, one for audio and the other for video - which we can ignore. The first titleset will often contain two titles, one for 2 channel audio and the other for 5.1 channel audio. Each title will usually contain a consistent number of tracks as MLP or PCM encoded audio.

With this in mind, we can now make some sense of the `AUDIO_TS` directory's contents:

`AUDIO_TS.IFO` information about the disc, including the number of titlesets

$\texttt{ATS\_}\underbrace{\texttt{01}}_{Titleset}\texttt{\_0.IFO}$ information about all the titles in a given titleset

$\texttt{ATS\_}\underbrace{\texttt{01}}_{Titleset}\texttt{\_}\underbrace{\texttt{1}}_{AOB\ \#}\texttt{.AOB}$ audio data for one or more tracks in a given titleset

All are binary files containing one or more, 2048 byte sectors.

17.1 AUDIO_TS.IFO

Known as the “Audio Manager” or “AMG”, this is primarily a container of pointers to other files on disc. However, for our purposes, we’re only interested in the ‘Audio Titleset Count’ value.

Identifier (`DVDAUDIO-AMG') 0 95			
Last AMG Sector 96 127	Padding 128 223	Last AMGI Sector 224 255	Padding 256 263
DVD Specification Version 264 271	Category 272 303		
Volume Count 304 319	Volume Number 320 335		
Disc Side 336 343	Padding 344 383	Unknown 384 415	Padding 416 495
Video Titleset Count 496 503	Audio Titleset Count 504 511	Provider Identifier 512 767	
...			

17.2 ATS_XX_0.IFO

Known as the “Audio Titleset Information” or “ATSI”, there is one of these files per ‘Audio Titleset Count’. Each is typically 4096 bytes long (2 sectors).

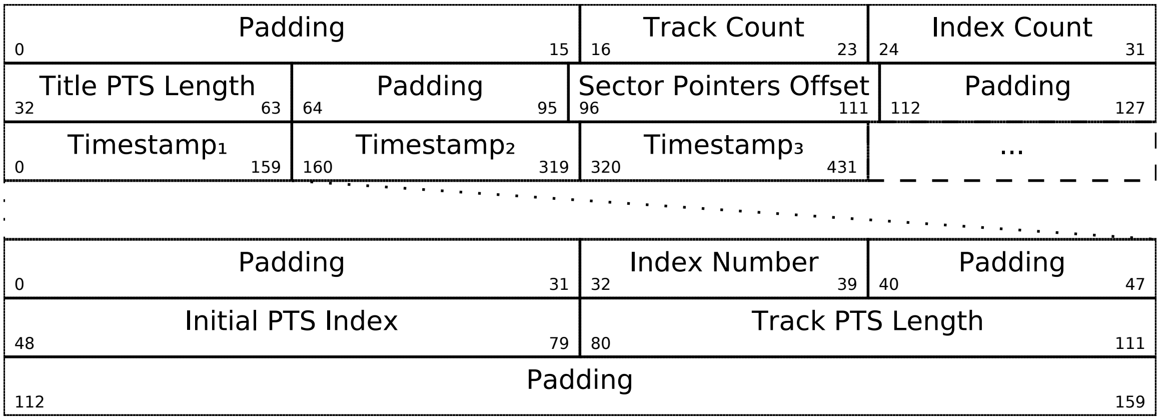
Identifier (`DVDAUDIO-ATS') 0 95	... 96 16383	
Title Count 16384 16399	Padding 16400 16415	Last Byte Address 16416 16447
Title Offset$_1$ 0 63	Title Offset$_2$ 64 127	...

Title Number 0 7	Padding 8 31	Title Table Offset 32 63

‘Title Table Offset’ is the address of the title’s information table, from the start of the second sector. For example, given the ‘Title Table Offset’ value of `0x18`, we seek to address `0x818` (`0x800` + `0x18` = `0x818`) to read the title’s information.

17.2.1 Title Table

For each title in the titleset, there is a title table.

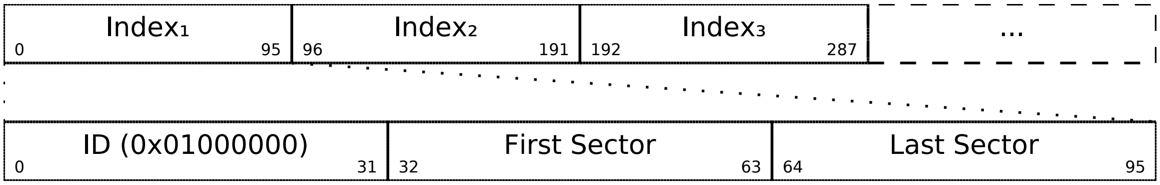


'Track Count' is the total number of tracks in the title. 'Title PTS Length' is the total length of the title in PTS ticks. There are 90000 PTS ticks per second. 'Sector Pointers Offset' is the offset value of the sector pointers table, relative to the start of the title table. For example, if our 'Title Table Offset' value is `0x18` and the 'Sector Pointers Offset' value is `0x1F0`, we seek to address `0xA08` to reach the sector pointers table (`0x818`+`0x1F0` = `0xA08`).

There is one timestamp value per track. 'Initial PTS Index' and 'Track PTS Length' values are the offset and length of the track in PTS ticks, respectively. 'Index Number' indicates the track's initial sector pointers values from the following table.

17.2.2 Sector Pointers Table

For each title in the titleset, there is a sector pointers table containing 'Index Count' number of 96 bit values.

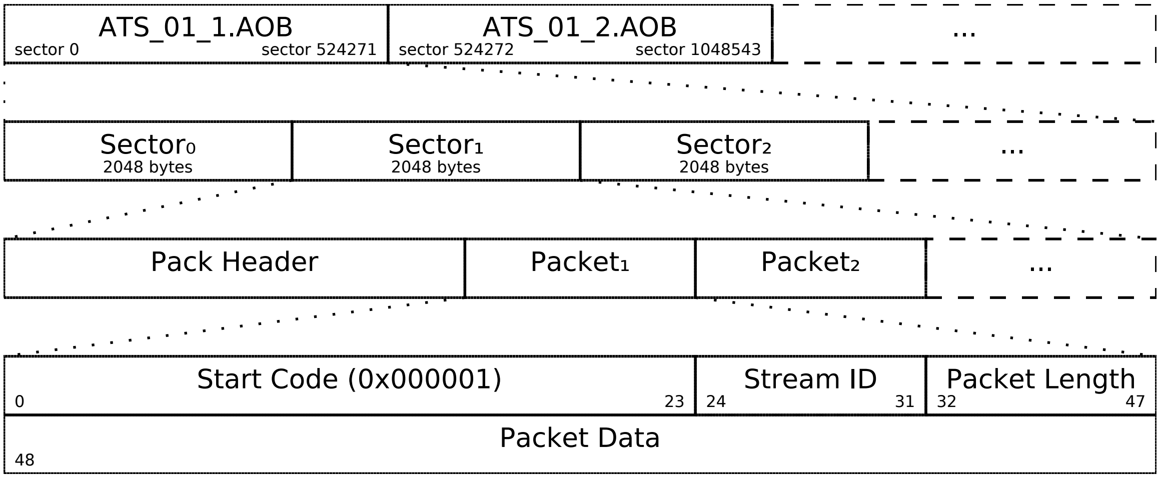


There is at least one sector pointers value per track. Each pointer is relative to the start of the titleset's first AOB file.

For example, if Track_1 has an 'Index Number' of 1 and Track_2 has an 'Index Number' of 5, that means Track_1 contains the sector pointers Index_1, Index_2, Index_3 and Index_4. However, much of the time 'Track Count' will equal 'Index Count'. In this case there will be only a single 'Index' value per track.

17.3 ATS_XX_X.AOB

All of a titleset's AOB files can be considered part of a single, contiguous collection of sectors, each 2048 bytes long. Thus, it's possible for the start and end sectors for a given track (as indicated in the sector pointers table) to span two or more AOB files. Each sector contains one or more packets as part of a "Packetized Elementary Stream".



Each sector within an AOB file is prefixed by a 'Pack Header', as follows:

Sync Bytes (0x000001BA) 0 31	0x1 32 33	Current PTS (high) 34 36	0x1 37	
Current PTS (mid) 38 52	0x1 53	Current PTS (low) 54 68	0x1 69	
SCR Extension 70 78	0x1 79	Bit Rate 80 101	0x3 102 103	
Reserved 104 108	Stuffing Count 109 111	Stuffing$_1$ 112 119	Stuffing$_2$ 120 127	...

The three 'Current PTS' values (3 bits, 15 bits and 15 bits, respectively) combine to indicate the current position within the stream, in PTS ticks.

Each sector then contains one or more packets. The packets with a 'Stream ID' of `0xBD` contain encoded audio data.

17.3.1 Audio Packet Data

Packets containining audio data have yet another header prior to the actual data. This header indicates the stream type (`0xA0` for raw PCM, `0xA1` for MLP). In the case of PCM, this header also contains details about the stream itself such as sample rate, bits per sample and channel assignment.

PCM Header

Unknown 0 15	Pad1 Size 16 23	Pad1 24 24 + (Pad1 Size × 8)	
Stream ID (0xA0) 0 7	CRC 8 15	Padding 16 23	Pad2 Size 24 31
First Audio Frame 32 47	Padding 48 55		
Group 1 Bits per Sample 56 59	Group 2 Bits per Sample 60 63		
Group 1 Sample Rate 64 67	Group 2 Sample Rate 68 71		
Padding 72 79	Channel Assignment 80 87	Padding 88 95	CRC 96 103
Padding 104 104 + (8 × (Pad2 Size - 9))	PCM Data		

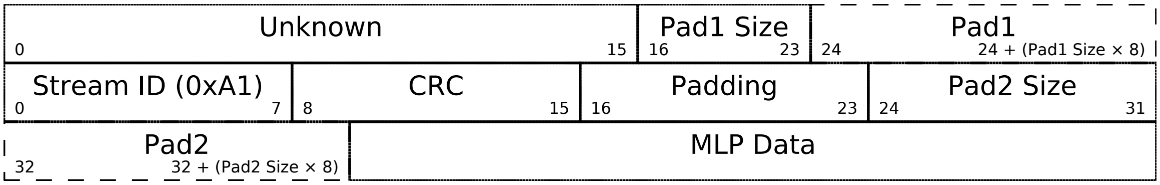
Value	Bits per Sample
0000	16
0001	20
0010	24
1111	Group Unused

Value	Sample Rate
0000	48000Hz
0001	96000Hz
0010	192000Hz
1000	44100Hz
1001	88200Hz
1010	176400Hz
1111	Group Unused

Value	Channel Assignment
00000	front center
00001	front left, front right
00010	front left, front right, back center
00011	front left, front right, back left, back right
00100	front left, front right, LFE
00101	front left, front right, LFE, back center
00110	front left, front right, LFE, back left, back right
00111	front left, front right, front center
01000	front left, front right, front center, back center
01001	front left, front right, front center, back left, back right
01010	front left, front right, front center, LFE
01011	front left, front right, front center, LFE, back center
01100	front left, front right, front center, LFE back left, back right
01101	front left, front right, front center, back center
01110	front left, front right, front center, back left, back right
01111	front left, front right, front center, LFE
10000	front left, front right, front center, LFE, back center
10001	front left, front right, front center, LFE, back left, back right
10010	front left, front right, back left, back right, LFE
10011	front left, front right, back left, back right, front center
10100	front left, front right, back left, back right, front center, LFE

Decoding the PCM data is explained on page 187.

MLP Header



Since the MLP stream contains details such as bits per sample and channel assignment, that information is not repeated in the MLP packet header.

Decoding the MLP data is explained on page 191.

17.4 PCM Decoding

For 16 bits-per-sample streams, PCM data is simply stored in signed, big-endian format.

However, for 24 bits-per-sample streams, the individual bytes must be reshuffled into a signed, little-endian depending on how many channels the stream has. This reshuffling occurs across *two* PCM frames. A 1 channel stream is reshuffled as follows:

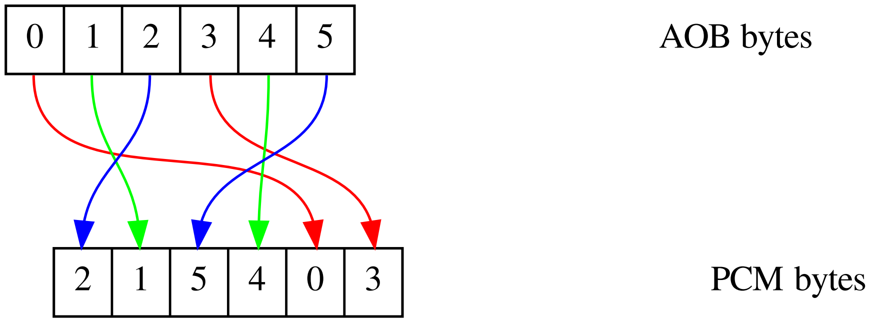


Figure 17.1: 1 channel, 24 bits-per-sample

For example, given the packet data bytes:

`EE BB CC DD FF AA`

we first reshuffle them to:

`CC BB AA FF EE DD`

We then transform those bytes into 2 PCM frames worth of 24-bit, little-endian samples:

$$\text{Frame}_1 = \texttt{0xAABBCC} = 2800588 - 2^{23} = -5588020$$
$$\text{Frame}_2 = \texttt{0xDDEEFF} = 6156031 - 2^{23} = -2232577$$

Streams with more than one channel are handled similarly.

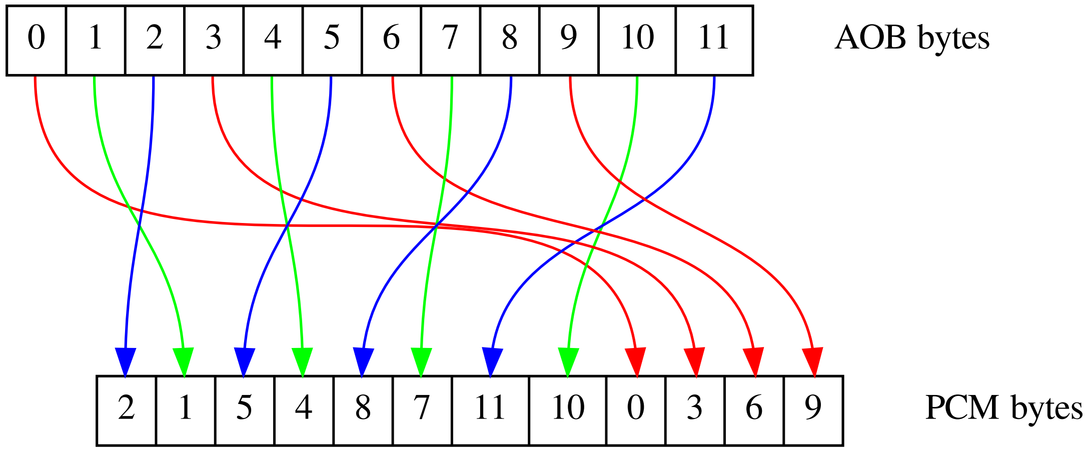


Figure 17.2: 2 channels, 24 bits-per-sample

For example, given the packet data bytes:

```
66 11 22 BB 55 00 AA 33 44 99 77 88
```

we first reshuffle them to:

```
22 11 00 55 44 33 88 77 66 BB AA 99
```

Then, we transform those bytes into 2 PCM frames worth of 24-bit, little-endian samples:

$$
\begin{aligned}
\text{Frame}_{1\ 1} &= \texttt{0x001122} = 4386\\
\text{Frame}_{1\ 2} &= \texttt{0x334455} = 3359829\\
\text{Frame}_{2\ 1} &= \texttt{0x667788} = 6715272\\
\text{Frame}_{2\ 2} &= \texttt{0x99AABB} = 1682107 - 2^{23} = -6706501
\end{aligned}
$$

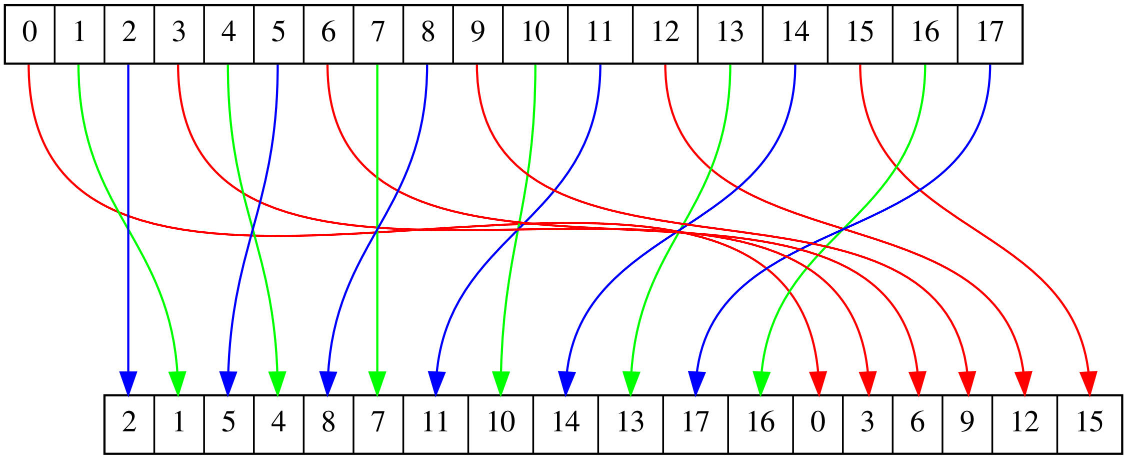


Figure 17.3: 3 channels, 24 bits-per-sample

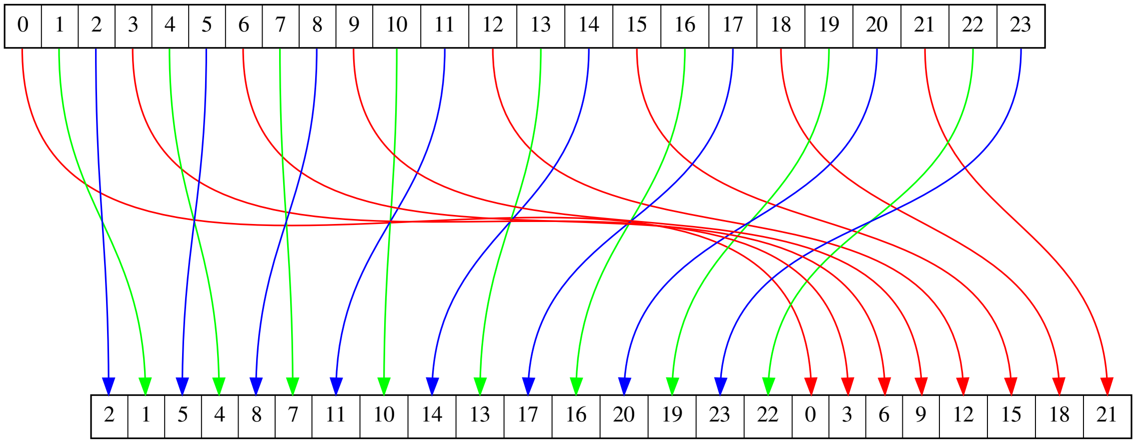


Figure 17.4: 4 channels, 24 bits-per-sample

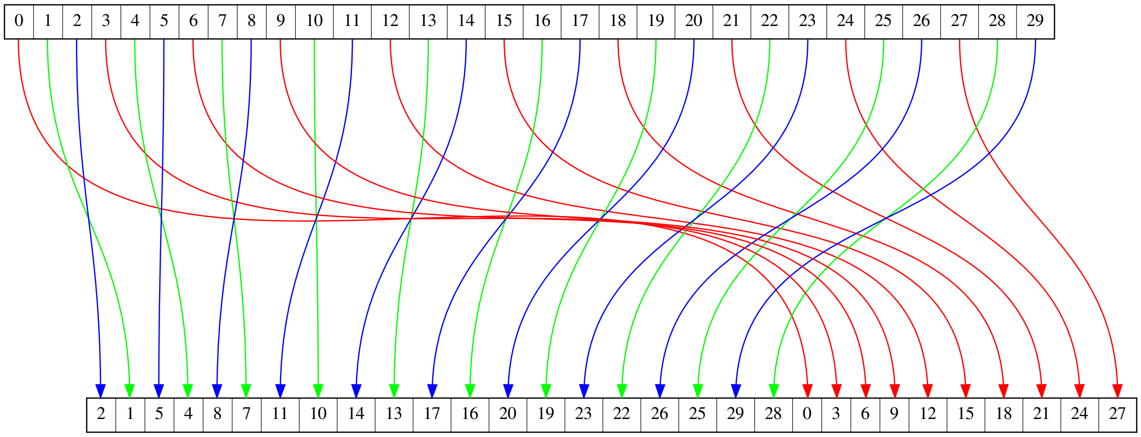


Figure 17.5: 5 channels, 24 bits-per-sample

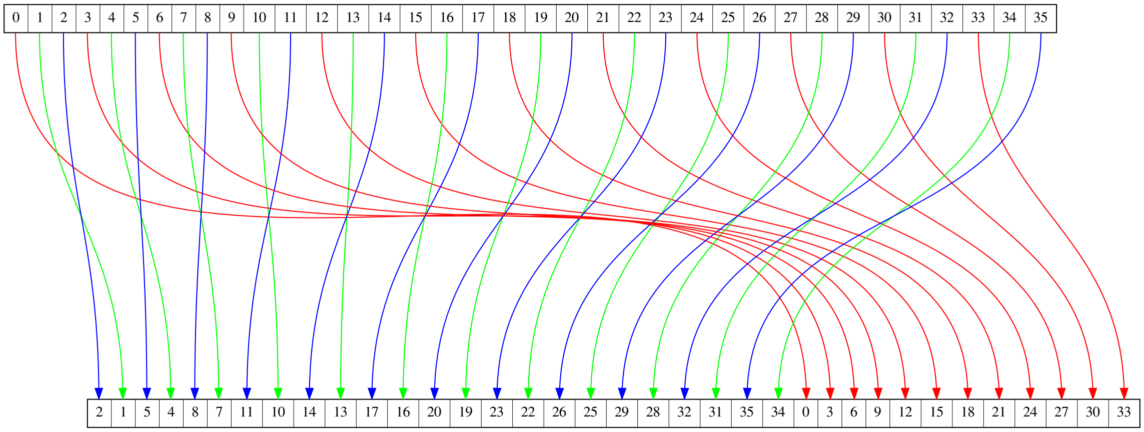
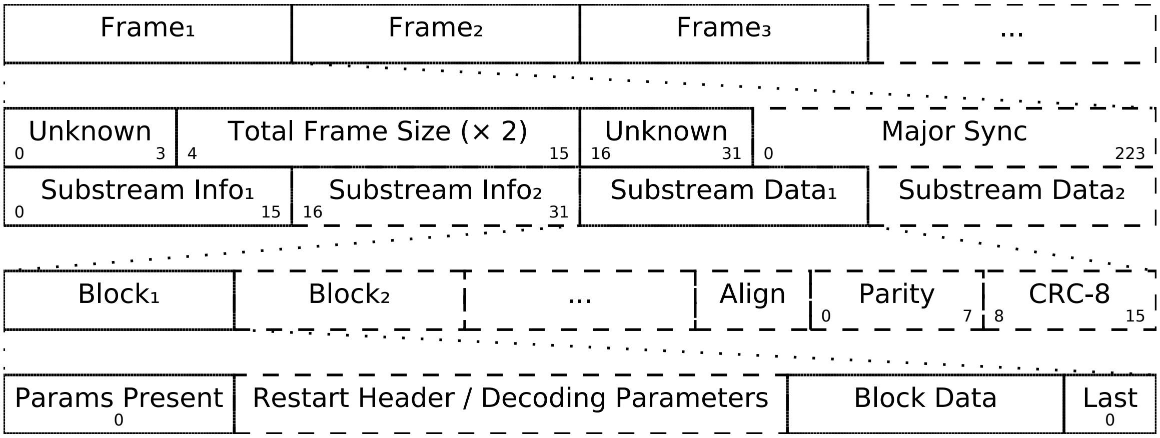


Figure 17.6: 6 channels, 24 bits-per-sample

17.5 Meridian Lossless Packing

An MLP stream is comprised of a series of frames. Each frame contains one or two substreams, each storing a different set of channels (e.g. Substream$_1$ may have channels 1-2, Substream$_2$ may have channels 3-6, and so on). Each substream contains one or more blocks. Finally, each block contains ‘Block Data’ residuals and, optionally, decoding information.



‘Total Frame Size’ is the size of the entire frame, including the header, in bytes.

The first frame will contain a ‘Major Sync’ with information such as sample rate, bits-per-sample, substream count and so forth. This information will be constant throughout the stream, so most of the subsequent MLP frames will not have a sync.

The basic principle of MLP decoding is that we read in a set of decoding parameters for the first block and use those as arguments for decoding the block data itself. Subsequent blocks may may contain truncated parameters indicating only which arguments have changed so that the next block can be decoded.

Decoded blocks are concatenated into a stream of PCM frames, each with one or more channels which comprise an entire substream.

The channels of all decoded substreams (each with an identical number of PCM frames) are then combined into a single set of PCM frames and returned.

17.5.1 The Major Sync

‘Major Sync’ is an optional frame header:

Sync Words (0xF8726F)	0	23
Stream Type (0xBB)	24	31
Group₁ Bits	32	35
Group₂ Bits	36	39
Group₁ Sample Rate	40	43
Group₂ Sample Rate	44	47
Unknown	48	58
Channel Assignment	59	63
Unknown	64	111
Is VBR	112	
Peak Bitrate	113	127
Substream Count	128	131
Unknown	132	223

We detect a ‘Major Sync’ by reading a 24-bit ‘Sync Words’ value and 8-bit ‘Stream Type’ value from the stream. If `0xF8726F` and `0xBB`, respectively, we can read the remainder of the sync. Otherwise, we must ‘back up’ 32 bits and continue reading substream end and substream data values from the stream. If a ‘Major Sync’ is not present in a frame, we use the values from the previous sync.

Value	Bits	Sample Rate	Channels	Channel Assignment
00000	16	48000	1	front center
00001	20	96000	2	front left, front right
00010	24	192000	3	front left, front right, back center
00011			4	front left, front right, back left, back right
00100			3	front left, front right, LFE
00101			4	front left, front right, LFE, back center
00110			5	front left, front right, LFE, back left, back right
00111			3	front left, front right, front center
01000		44100	4	front left, front right, front center, back center
01001		88200	5	front left, front right, front center, back left, back right
01010		176400	4	front left, front right, front center, LFE
01011			5	front left, front right, front center, LFE, back center
01100			6	front left, front right, front center, LFE back left, back right
01101			4	front left, front right, front center, back center
01110			5	front left, front right, front center, back left, back right
01111			4	front left, front right, front center, LFE
10000			5	front left, front right, front center, LFE, back center
10001			6	front left, front right, front center, LFE, back left, back right
10010			5	front left, front right, back left, back right, LFE
10011			5	front left, front right, back left, back right, front center
10100			6	front left, front right, back left, back right, front center, LFE

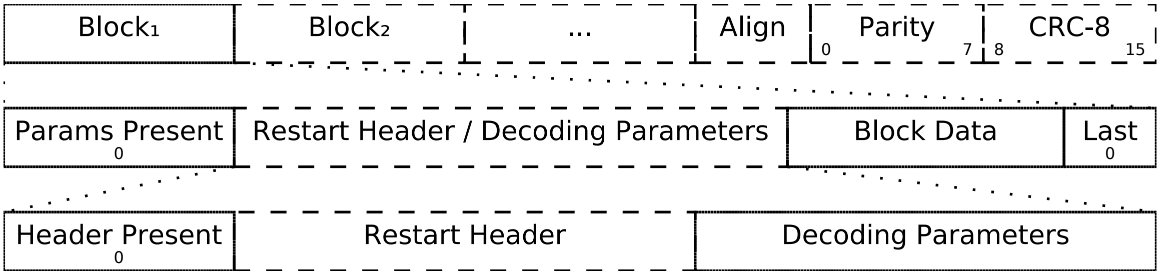
17.5.2 Substream Info

There is one ‘Substream Info’ header per frame substream (as indicated in the ‘Major Sync’).

Extraword Present (0)	0	
Nonrestart Substream	1	
Checkdata Present	2	
Skip	3	
Substream End (× 2)	4	15

‘Substream End’ is the location of the substream’s end, starting from the end of the last ‘Substream Info’ value.

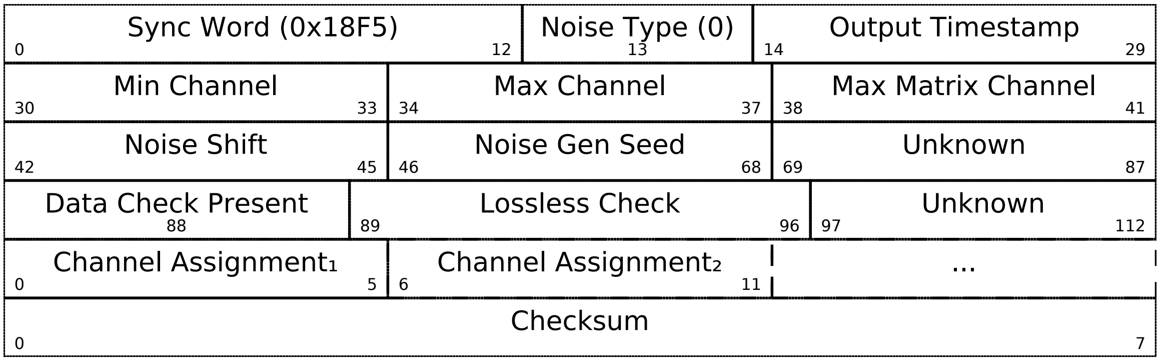
17.5.3 Substream Data



We continue to read blocks until a block's 'Last' bit is set. Each block is prefixed by a 'Parameters Present' bit. If set, the block has an optional 'Restart Header' block (indicated by the prefixed 'Header Present' bit) and required 'Decoding Parameters' block. If 'Parameters Present' is not set, we continue directly to 'Block Data' - using the most recently read set of parameters for decoding.

Once all the blocks have been read, we align the stream to a mulitple of 16-bits. Then, if 'Checkdata Present' has been indicated in the 'Substream Info' header, we read an 8-bit parity and an 8-bit CRC value.

17.5.4 Restart Header

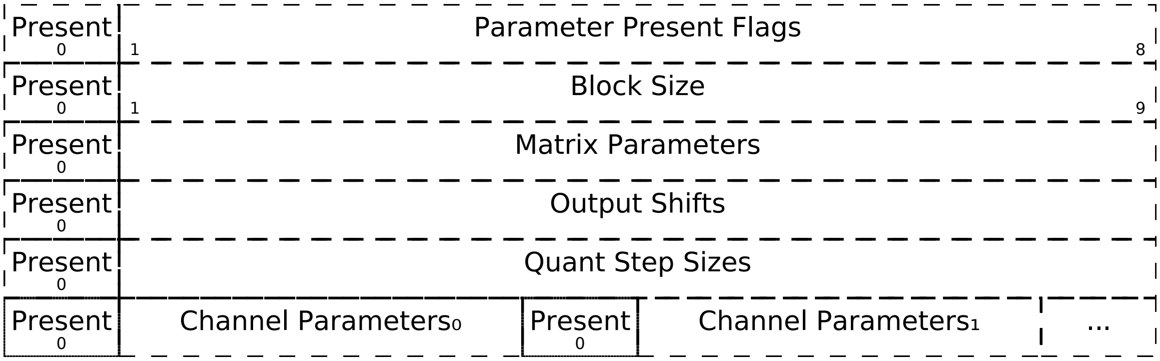


'Min Channel' and 'Max Channel' indicate the substream's range of channels. For example, a 6 channel stream might have values of 0 and 1 for the first substream, and values of 2 and 5 for the second. So, decoding substream$_1$ would generate sample$_{0\ i}$ and sample$_{1\ i}$, while decoding substream$_2$ would generate sample$_{2\ i}$ through sample$_{5\ i}$ (where i covers the range of total samples in the substreams' blocks).

There are 'Max Matrix Channel' + 1 number of 'Channel Assignment' values, indicating which MLP channel the decoded output will be sent to (offset from the substream's minimum channel).

17.5.5 Decoding Parameters

Reading the decoding parameters requires 8, ‘Parameter Presence Flags’ bits. If the flag for a particular parameter is set, *and* the parameter’s prefixed ‘Present’ bit is set, we go ahead and read that parameter. Note that the first decoding parameter is the 8-bit ‘Parameter Presence Flags’ value itself.



There is one optional ‘Channel Parameters’ block per channel from ‘Min Channel’ to ‘Max Channel’. Again, each is prefixed by a single ‘Present’ bit.

For example, given the our 2 substream MLP file from before (with ‘Min Channel’ = 0, ‘Max Channel’ = 1 in substream$_1$ and ‘Min Channel’ = 2, ‘Max Channel’ = 5 in substream$_2$), the first substream has optional Channel Parameters$_0$ and Channel Parameters$_1$, while the second has optional Channel Parameters$_2$ through Channel Parameters$_5$.

Parameter Presence Flags

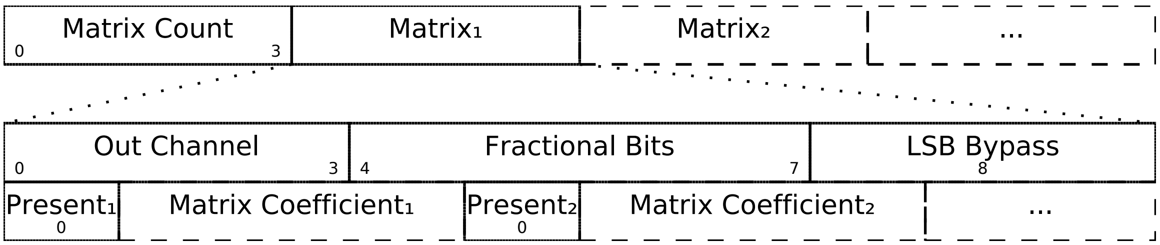
Presence Flags 0	Huffman Offset 1	IIR Filter Parameters 2	FIR Filter Parameters 3
Quant Step Sizes 4	Output Shifts 5	Matrix Parameters 6	Block Size 7

By default, all 8 flags are all set. The ‘Huffman Offset’, ‘IIR Filter Parameters’ and ‘FIR Filter Parameters’ flags are checked while reading ‘Channel Parameters’.

Block Size

This is the size of the block in PCM frames. Its value is 8 by default. All channels within a block will have the same number of samples.

Matrix Parameters

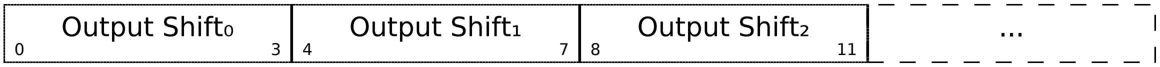


There are ‘Matrix Count’ number of matrices. There is one coefficient per ‘Max Matrix Channel’ + 3. The size of each signed matrix coefficient is ‘Fractional Bits’ + 2 and its value is calculated as:

$$\text{Matrix Coefficient}_i = \text{Signed Value}_i \times 2^{14-\text{Fractional Bits}}$$

If the preceding ‘Present’ bit is *not* set, the value of Matrix Coefficient$_i$ is 0. If ‘Matrix Parameters’ is *not* indicated during parameter decoding, treat ‘Matrix Count’ as 0 by default.

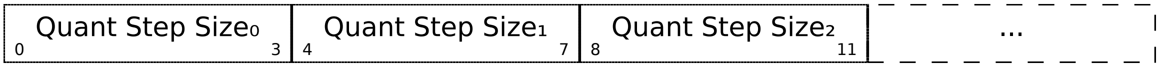
Output Shifts



There are ‘Max Matrix Channel’ + 1 number of shift values. Each is a signed, 4-bit value.

If ‘Output Shifts’ is *not* indicated during parameter decoding, all Output Shift$_i$ values are 0 by default.

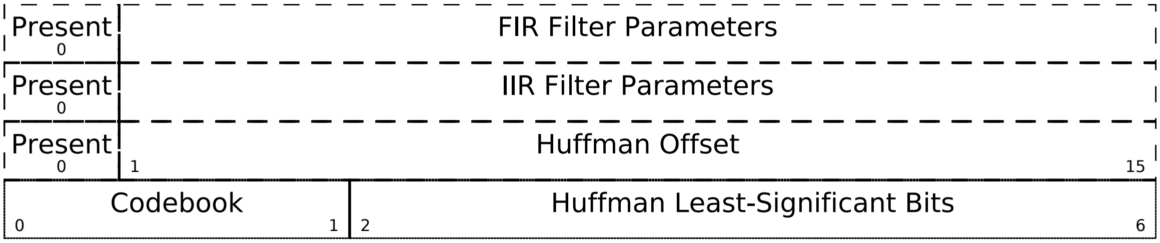
Quant Step Sizes



There is one ‘Quant Step Size’ value per substream channel. To continue our 2 substream example, if the ‘Present’ bit is set, the first substream has Quant Step Size$_0$ and Quant Step Size$_1$, while the second has Quant Step Size$_2$ through Quant Step Size$_5$.

Each is an unsigned, 4-bit value. These are analagous to FLAC’s wasted bits-per-sample. If ‘Quant Step Sizes’ is *not* indicated during parameter decoding, all Quant Step Size$_i$ values are 0 by default.

Channel Parameters



Remember that the optional ‘FIR Filter Parameters’, ‘IIR Filter Parameters’ and ‘Huffman Offset’ values are read only if the corresponding ‘Parameter Presence Flag’ is set.

‘Codebook’ and ‘Huffman Least-Significant Bits’ are unsigned values. ‘Huffman Offset’ is a signed value. ‘Signed Huffman Offset’ is calculated as follows:

If Codebook > 0:

$$\begin{aligned} \text{LSB Bits} &= \text{Huffman Least-Significant Bits} - \text{Quant Step Size}_{\text{channel}} \\ \text{Sign Shift} &= \text{LSB Bits} + 2 - \text{Codebook} \\ \text{Signed Huffman Offset} &= \begin{cases} \text{Huffman Offset} - 7 \times 2^{\text{LSB Bits}} - 2^{\text{Sign Shift}} & \text{if Sign Shift} \geq 0 \\ \text{Huffman Offset} - 7 \times 2^{\text{LSB Bits}} & \text{if Sign Shift} < 0 \end{cases} \end{aligned}$$

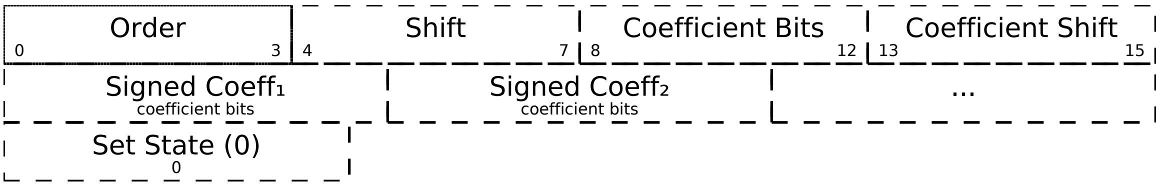
If Codebook $= 0$:

$$\begin{aligned} \text{LSB Bits} &= \text{Huffman Least-Significant Bits} - \text{Quant Step Size}_{\text{channel}} \\ \text{Sign Shift} &= \text{LSB Bits} - 1 \\ \text{Signed Huffman Offset} &= \begin{cases} \text{Huffman Offset} - 2^{\text{Sign Shift}} & \text{if Sign Shift} \geq 0 \\ \text{Huffman Offset} & \text{if Sign Shift} < 0 \end{cases} \end{aligned}$$

The ‘Signed Huffman Offset’ will be added to each decoded residual during block decoding.

By default, ‘Signed Huffman Offset’ is -2^{23}, ‘Codebook’ is 0 and ‘Huffman Least-Significant Bits’ is 23.

FIR Filter Parameters



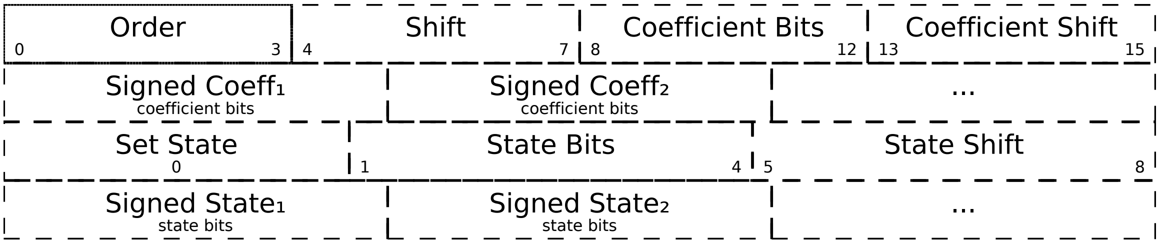
If ‘Order’ is 0, we forego reading the rest of the filter parameters. Otherwise, ‘Shift’, ‘Coefficient Bits’ and ‘Coefficient Shift’ are unsigned values - the latter two are used for reading ‘Order’ number of signed values which we convert to filter coefficients as follows:

$$\text{FIR Filter Coefficient}_i = \text{Signed Coeff}_i \times 2^{\text{Coefficient Shift}}$$

We’re primarily interested in the ‘Shift’ and ‘FIR Filter Coefficient’ values, which will be used for filtering decoded residuals.

Note that the ‘Set State’ value must be 0. By default, ‘Shift’ is 0 and the signed coefficient list is empty.

IIR Filter Parameters



As with FIR, if ‘Order’ is 0, we forego reading the rest of the filter parameters. Otherwise, ‘Shift’, ‘Coefficient Bits’ and ‘Coefficient Shift’ are unsigned values - the latter two are used for reading ‘Order’ number of signed coefficient values which we convert to filter coefficients as follows:

$$\text{IIR Filter Coefficient}_i = \text{Signed Coeff}_i \times 2^{\text{Coefficient Shift}}$$

If ‘Set State’ is 1, we also read ‘Order’ number of signed state values which we convert as follows:

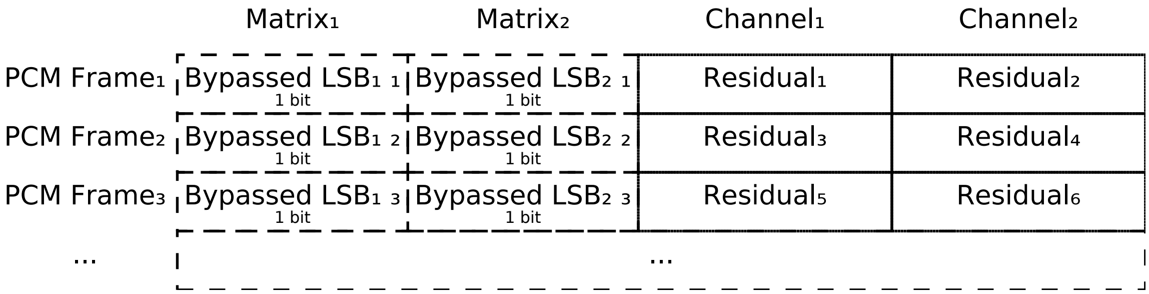
$$\text{State}_i = \text{Signed State}_i \times 2^{\text{State Shift}}$$

We’re primarily interested in the ‘Shift’ and ‘IIR Filter Coefficient’ and ‘State’ values which will be used for filtering decoded residuals.

By default, ‘Shift’ is 0 and the signed coefficient list is empty.

17.5.6 Block Data

The block data of a given substream block is the residuals from which our original samples will be reconstructed. Residuals are interleaved between substream channels. That is, given a 2 channel substream, Residual$_1$ is on Channel$_1$, Residual$_2$ is on Channel$_2$, Residual$_3$ is on Channel$_1$ and so forth. Furthermore, for each 'Matrix Parameter' in the substream, if 'LSB Bypass' is set in that parameter, one must read a 'Bypassed LSB' bit per PCM frame's worth of residuals prior to reading the residuals themselves.



	Matrix$_1$	Matrix$_2$	Channel$_1$	Channel$_2$
PCM Frame$_1$	Bypassed LSB$_{1\ 1}$ 1 bit	Bypassed LSB$_{2\ 1}$ 1 bit	Residual$_1$	Residual$_2$
PCM Frame$_2$	Bypassed LSB$_{1\ 2}$ 1 bit	Bypassed LSB$_{2\ 2}$ 1 bit	Residual$_3$	Residual$_4$
PCM Frame$_3$	Bypassed LSB$_{1\ 3}$ 1 bit	Bypassed LSB$_{2\ 3}$ 1 bit	Residual$_5$	Residual$_6$
...		...		

Decoding a given residual value requires the 'Codebook', 'Huffman Least-Significant Bits' and 'Signed Huffman Offset' values from its 'Channel Parameters', as well as the 'Quant Step Size' for that residual's channel. First, given a positive 'Codebook' value, we decode a Huffman value from the bit stream as indiciated by the trees on page 199. This value is our 'Most-Signicant Bits' (MSB). If 'Codebook' is 0, our 'Most-Significant Bits' are also 0.

We then calculate our least-significant bit count as:

$$\text{LSB Count}_c = \text{Huffman Least-Significant Bits}_c - \text{Quant Step Size}_c$$

We read this number of bits from the stream as our 'Least-Significant Bits' (LSB) value. If LSB Count$_c$ is 0, we read no bits from the stream and our 'Least-Significant Bits' value is also 0.

Our final residual value is calculated as follows:

$$\text{Residual}_i = ((\text{MSB}_i \times 2^{\text{LSB Count}_i}) + \text{LSB}_i + \text{Signed Huffman Offset}_c) \times 2^{\text{Quant Step Size}_c}$$

Note that if 'Codebook' is 0 *and* our LSB Count$_c$ is 0, we read no bits from the stream and Residual$_i$ is 0. This is how MLP stores large runs of 0 residuals.

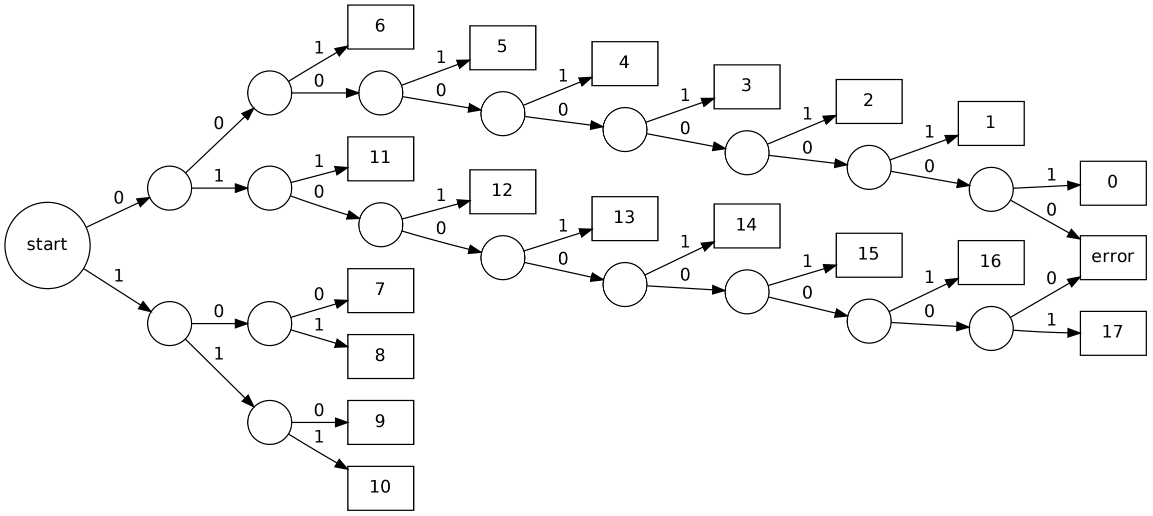


Figure 17.7: Codebook 1

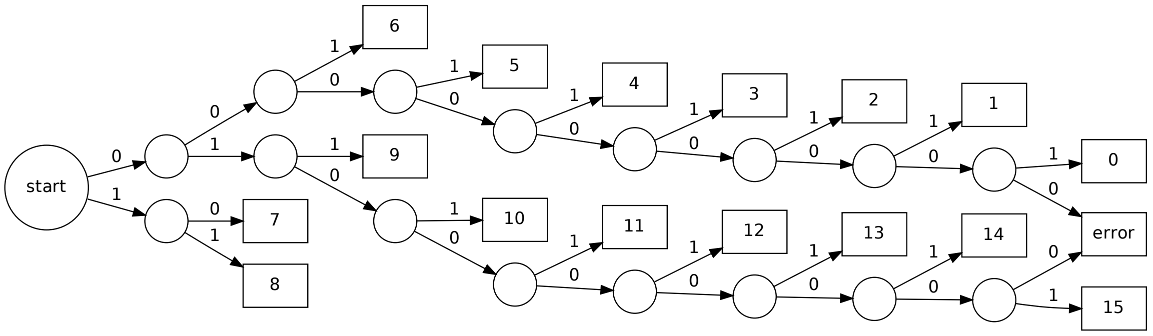


Figure 17.8: Codebook 2

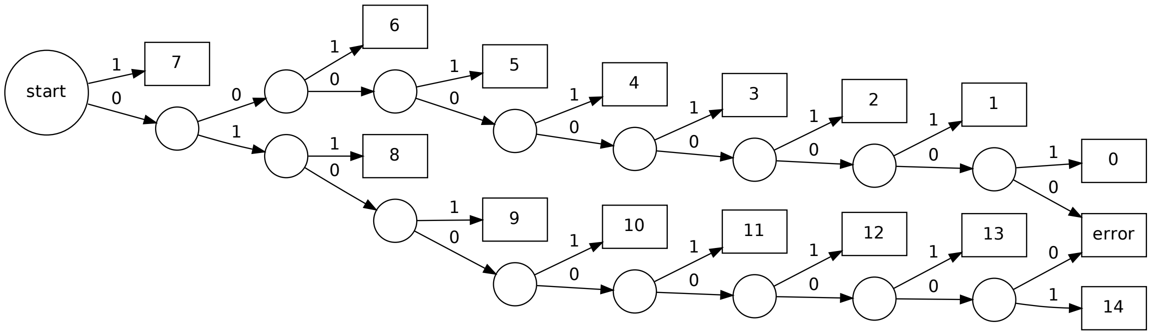


Figure 17.9: Codebook 3

17.5.7 Channel Filtering

Given a channel's decoded block residuals, we then filter those residuals with the channel's FIR and IIR filters from its 'Decoding Parameters'. In particular, we'll need the 'FIR Order', 'FIR Coefficients', 'IIR Order', 'IIR Coefficients', 'IIR State' and 'Shift'[1] parameters, along with the 'Quant Step Size' for the filtered channel.

Note that the 'Filtered' values below 0 are taken from the previously filtered block.

Quant Step Size for a given channel is used to build a mask function which zeroes out least-significant bits:

$$\text{mask(x)} = \lfloor x \div 2^{\text{Quant Step Size}_c} \rfloor \times 2^{\text{Quant Step Size}_c}$$

The filtered values are then calculated as follows:

$$\begin{aligned}
\text{FIR Sum}_i &= \sum_{j=0}^{\text{FIR Order}-1} \text{FIR Coefficient}_j \times \text{Filtered}_{i-j-1} \\
\text{IIR Sum}_i &= \sum_{k=0}^{\text{IIR Order}-1} \text{IIR Coefficient}_k \times \text{IIR State}_{i-k-1+\text{IIR Order}} \\
\text{Shifted Sum}_i &= \left\lfloor \frac{\text{FIR Sum}_i + \text{IIR Sum}_i}{2^{\text{Shift}}} \right\rfloor \\
\text{Filtered}_i &= \text{mask}(\text{Shifted Sum}_i + \text{Residual}_i) \\
\text{IIR State}_{i+\text{IIR Order}} &= \text{Filtered}_i - \text{Shifted Sum}_i
\end{aligned}$$

However, if both 'FIR Order' and 'IIR Order' are 0, this can be reduced to:

$$\text{Filtered}_i = \text{mask}(\text{Residual}_i)$$

Also note that if Quant Step Size$_c = 0$:

$$\begin{aligned}
\text{mask}(x) &= \lfloor x \div 2^0 \rfloor \times 2^0 = x \\
\text{IIR State}_{i+\text{IIR Order}} &= \text{mask}(\text{Shifted Sum}_i + \text{Residual}_i) - \text{Shifted Sum}_i \\
&= \text{Residual}_i
\end{aligned}$$

[1] If both both FIR filter and IIR filter have a positive number of coefficients, their 'Shift' values must be identical

As an example, given the values:

FIR Order	=	4	IIR Order	=	3	Filtered$_{-4}$	=	-40
FIR Coefficient$_0$	=	556	IIR Coefficient$_0$	=	-122	Filtered$_{-3}$	=	-39
FIR Coefficient$_1$	=	-493	IIR Coefficient$_1$	=	102	Filtered$_{-2}$	=	0
FIR Coefficient$_2$	=	288	IIR Coefficient$_2$	=	-192	Filtered$_{-1}$	=	18
FIR Coefficient$_3$	=	-96	IIR State$_0$	=	12	Residual$_0$	=	34
Shift	=	8	IIR State$_1$	=	20	Residual$_1$	=	9
Quant Step Size$_c$	=	0	IIR State$_2$	=	-8	Residual$_2$	=	-23

Our filtered values are as follows:

$$\begin{aligned}
\text{FIR Sum}_0 &= (556 \times 18) + (-493 \times 0) + (288 \times -39) + (-96 \times -40) = 2616 \\
\text{IIR Sum}_0 &= (-122 \times 12) + (102 \times 20) + (-192 \times -8) = 2112 \\
\text{Shifted Sum}_0 &= \left\lfloor \frac{2616 + 2112}{2^8} \right\rfloor = 18 \\
\text{Filtered}_0 &= \text{mask}(18 + 34) = \mathbf{52} \\
\text{IIR State}_3 &= 52 - 18 = 34 \\
\text{FIR Sum}_1 &= (556 \times 52) + (-493 \times 18) + (288 \times 0) + (-96 \times -39) = 23782 \\
\text{IIR Sum}_1 &= (-122 \times 34) + (102 \times 12) + (-192 \times 20) = -6764 \\
\text{Shifted Sum}_1 &= \left\lfloor \frac{23782 - 6764}{2^8} \right\rfloor = 66 \\
\text{Filtered}_1 &= 66 + 9 = \mathbf{75} \\
\text{IIR State}_4 &= 75 - 66 = 9 \\
\text{FIR Sum}_2 &= (556 \times 75) + (-493 \times 52) + (288 \times 18) + (-96 \times 0) = 21248 \\
\text{IIR Sum}_2 &= (-122 \times 9) + (102 \times 34) + (-192 \times 12) = 66 \\
\text{Shifted Sum}_2 &= \left\lfloor \frac{21248 + 66}{2^8} \right\rfloor = 83 \\
\text{Filtered}_2 &= 83 - 23 = \mathbf{60} \\
\text{IIR State}_5 &= 60 - 83 = -23
\end{aligned}$$

17.5.8 Channel Rematrixing

Finally, given our filtered channel values and a set of matrix parameters, we “rematrix” those values into the final PCM values. This is a form of cross-channel decorrelation, roughly analagous to FLAC’s mid-side channel assignment.

Note that if a file has two substreams, we apply matrix parameters depending on how many substreams are decoded. That is, when decoding only the first substream, we only apply the first substream’s matrix parameters. But when decoding both substreams, we apply the second substream’s matrix parameters. This allows a two substream MLP file to have two different “mixes” from the same data.

Noise Channels

First, we generate two noise channels, each containing ‘Block Size’ number of noise samples, using the ‘Noise Gen Seed’ and ‘Noise Shift’ (all taken from the current ‘Restart Header’). This requires a simple crop function which converts an integer to a signed, 8-bit value:

$$\text{crop}(x) = \begin{cases} x \bmod 2^8 & \text{if } \lfloor (x \bmod 2^8) \div 2^7 \rfloor = 0 \\ (x \bmod 2^7) - 2^7 & \text{if } \lfloor (x \bmod 2^8) \div 2^7 \rfloor \neq 0 \end{cases}$$

$$\begin{aligned}
\text{Shifted}_i &= \lfloor \text{Seed}_i \div 2^7 \rfloor \bmod 2^{16} \\
\text{Noise}_{i\ 1} &= \text{crop}(\lfloor \text{Seed}_i \div 2^{15} \rfloor) \times 2^{\text{Noise Shift}} \\
\text{Noise}_{i\ 2} &= \text{crop}(\text{Shifted}_i) \times 2^{\text{Noise Shift}} \\
\text{Seed}_{i+1} &= ((\text{Seed}_i \times 2^{16}) \bmod 2^{32}) \textbf{ xor } \text{Shifted}_i \textbf{ xor } (\text{Shifted}_i \times 2^5)
\end{aligned}$$

For example, given a ‘Noise Gen Seed’ of 7855724 and a ‘Noise Shift’ of 0, our first couple of noise samples are as follows:

$$\begin{aligned}
\text{Seed}_0 &= 7855724 \\
\text{Shifted}_0 &= \lfloor 7855724 \div 2^7 \rfloor \bmod 2^{16} = 61372 \\
\text{Noise}_{0\ 1} &= \text{crop}(\lfloor 7855724 \div 2^{15} \rfloor) \times 2^0 = \text{crop}(239) \times 1 = \textbf{-17} \\
\text{Noise}_{0\ 2} &= \text{crop}(61372) \times 2^0 = \textbf{-68} \\
\text{Seed}_1 &= ((7855724 \times 2^{16}) \bmod 2^{32}) \textbf{ xor } 61372 \textbf{ xor } 61372 \times 2^5 \\
&= 3731619840 \textbf{ xor } 61372 \textbf{ xor } 1963904 = 3731953724 \\
\text{Shifted}_1 &= \lfloor 3731953724 \div 2^7 \rfloor \bmod 2^{16} = 57904 \\
\text{Noise}_{1\ 1} &= \text{crop}(\lfloor 3731953724 \div 2^{15} \rfloor) \times 2^1 = \text{crop}(113890) \times 1 = \textbf{-30} \\
\text{Noise}_{1\ 2} &= \text{crop}(57904) \times 2^0 = \textbf{48} \\
\text{Seed}_2 &= ((3731953724 \times 2^{16}) \bmod 2^{32}) \textbf{ xor } 57904 \textbf{ xor } 57904 \times 2^5 \\
&= 406585344 \textbf{ xor } 57904 \textbf{ xor } 1852928 = 404792368
\end{aligned}$$

Rematrixing

Performing the actual rematrixing requires appending the two noise channels to the end of each PCM frame of filtered residual data. That is, for a 2 channel stream:

$$\begin{array}{rcr}
\text{Matrix}_{i\ 1} & = & \text{Filtered}_{i\ 1} \\
\text{Matrix}_{i\ 2} & = & \text{Filtered}_{i\ 2} \\
\text{Matrix}_{i\ 3} & = & \text{Noise}_{i\ 1} \\
\text{Matrix}_{i\ 4} & = & \text{Noise}_{i\ 2}
\end{array}$$

The calculation for the matrix's output sample is then as follows:

$$\text{Shifted Sum}_{i\ \text{Channel}} = \left\lfloor \frac{\sum_{j=1}^{\text{Channel Count} + 2} \text{Matrix}_{i\ j} \times \text{Matrix Coefficient}_j}{2^{14}} \right\rfloor$$

$$\text{Rematrixed}_{i\ \text{Channel}} = \text{mask}(\text{Shifted Sum}_{i\ \text{Channel}}) + \text{Bypassed LSB}_i$$

Note that the two noise channels are the reason we read 'Max Matrix Channel' + 3 number of matrix coefficients instead of one coefficient per channel as one might expect. Also note that the 'Bypassed LSB' bit we read in the 'Block Data' for each PCM frame reappears at this point. The mask function is the one used for channel filtering on page 200. So, given the following values:

$$\begin{array}{rcrrcr}
\text{Filtered}_{0\ 1} & = & -54372 & \text{Matrix Coefficient}_1 & = & -16384 \\
\text{Filtered}_{0\ 2} & = & 169057 & \text{Matrix Coefficient}_2 & = & 6144 \\
\text{Noise}_{0\ 1} & = & -17 & \text{Matrix Coefficient}_3 & = & 0 \\
\text{Noise}_{0\ 2} & = & -68 & \text{Matrix Coefficient}_4 & = & 0 \\
\text{Filtered}_{1\ 1} & = & -92558 & \text{Bypassed LSB}_0 & = & 0 \\
\text{Filtered}_{1\ 2} & = & 118830 & \text{Bypassed LSB}_1 & = & 0 \\
\text{Noise}_{1\ 1} & = & -30 & \text{Channel Count} & = & 2 \\
\text{Noise}_{1\ 2} & = & -48 & \text{Out Channel} & = & 0
\end{array}$$

Our new values for channel 1 ('Out Channel' + 1) are calculated as follows:

$$\begin{aligned}
\text{Rematrixed}_{0\ 1} &= \left\lfloor \frac{(-54372 \times -16384) + (169057 \times 6144) + (-17 \times 0) + (-68 \times 0)}{2^{14}} \right\rfloor + 0 \\
&= \left\lfloor \frac{1929517056}{16384} \right\rfloor = \mathbf{117768} \\
\text{Rematrixed}_{1\ 1} &= \left\lfloor \frac{(-92558 \times -16384) + (118830 \times 6144) + (-30 \times 0) + (-48 \times 0)}{2^{14}} \right\rfloor + 0 \\
&= \left\lfloor \frac{2246561792}{16384} \right\rfloor = \mathbf{137119}
\end{aligned}$$

The values for channel 2 remain unchanged as 169057 and 118830, respectively.

17.5.9 Output Shifts

If ‘Output Shifts’ are indicated in the decoding parameters, one applies them to the rematrixed samples on a per-channel basis as follows:

$$\text{Final Sample}_{i\ \text{Channel}} = \text{Rematrixed}_{i\ \text{Channel}} \times 2^{\text{Output Shift}_{\text{Channel}}}$$

Naturally, when ‘Output Shift’ is 0 for a given channel, the rematrixed samples are our final samples.

Once the final block has been read (as indicated by its ‘Last’ bit), we align the stream to a multiple of 16 bits before checking for an end of stream marker or optional parity/CRC-8 bytes.

17.5.10 The End-of-Stream Marker

The MLP optional end-of-stream marker is indicated by 4 bytes:

```
D2 34 D2 34
```

which are found *after* byte alignment but *prior* to the optional parity and CRC-8 bytes. Not all MLP streams have an end-of-stream marker at the actual end of the stream, nor is there any guarantee that a false end-of-stream marker won’t appear at the end of block data. Therefore, one should only check for an optional marker once the stream’s samples have been exhausted - as indicated by the ‘Track PTS Length’ field in the title table, described on page 183.

17.5.11 Parity

If ‘Checkdata Present’ is indicated in the ‘Restart Header’, each substream is followed by a parity and CRC byte. Calculating the parity value is a simple matter of performing **xor** on all the substream’s bytes including the parity byte itself. If successful, the result should be `0xA9`. In mathematical terms, this calculation is as follows:

$$\texttt{0xA9} = \text{Parity Byte } \mathbf{xor} \text{ parity(Substream Size} - 2)$$

Where the parity function is defined as:

$$\text{parity}(x) = \begin{cases} \text{Substream Byte}_0 & \text{if } x = 0 \\ \text{Substream Byte}_x \ \mathbf{xor} \ \text{parity}(x-1) & \text{if } x > 0 \end{cases}$$

For example, given the substream bytes `0x95 0x00 0x20 0x00 0x1C 0xB6`, our parity byte is `0x1C` and the parity calculation is as follows:

$$\text{Parity} = \texttt{0x1C} \ \mathbf{xor} \ \texttt{0x00} \ \mathbf{xor} \ \texttt{0x20} \ \mathbf{xor} \ \texttt{0x00} \ \mathbf{xor} \ \texttt{0x95} = \texttt{0xA9}$$

17.5.12 CRC-8

Calculating the CRC-8 value is similar:

$$\text{CRC-8 Byte} = \text{crc}(\text{Substream Size} - 2)$$

Where the crc function is defined as:

$$\text{crc}(x) = \begin{cases} \text{Substream Byte}_x \textbf{ xor } \text{crc}(x-1) & \text{if } x = \text{Substream Size} - 2 \\ \text{CRC8}(\text{Substream Byte}_x \textbf{ xor } \text{crc}(x-1)) & \text{if } x > 0 \\ \text{CRC8}(\text{Substream Byte}_0 \textbf{ xor } \texttt{0x3C}) & \text{if } x = 0 \end{cases}$$

and CRC8 is taken from the following table:

	0x?0	0x?1	0x?2	0x?3	0x?4	0x?5	0x?6	0x?7	0x?8	0x?9	0x?A	0x?B	0x?C	0x?D	0x?E	0x?F
0x0?	0x00	0x63	0xC6	0xA5	0xEF	0x8C	0x29	0x4A	0xBD	0xDE	0x7B	0x18	0x52	0x31	0x94	0xF7
0x1?	0x19	0x7A	0xDF	0xBC	0xF6	0x95	0x30	0x53	0xA4	0xC7	0x62	0x01	0x4B	0x28	0x8D	0xEE
0x2?	0x32	0x51	0xF4	0x97	0xDD	0xBE	0x1B	0x78	0x8F	0xEC	0x49	0x2A	0x60	0x03	0xA6	0xC5
0x3?	0x2B	0x48	0xED	0x8E	0xC4	0xA7	0x02	0x61	0x96	0xF5	0x50	0x33	0x79	0x1A	0xBF	0xDC
0x4?	0x64	0x07	0xA2	0xC1	0x8B	0xE8	0x4D	0x2E	0xD9	0xBA	0x1F	0x7C	0x36	0x55	0xF0	0x93
0x5?	0x7D	0x1E	0xBB	0xD8	0x92	0xF1	0x54	0x37	0xC0	0xA3	0x06	0x65	0x2F	0x4C	0xE9	0x8A
0x6?	0x56	0x35	0x90	0xF3	0xB9	0xDA	0x7F	0x1C	0xEB	0x88	0x2D	0x4E	0x04	0x67	0xC2	0xA1
0x7?	0x4F	0x2C	0x89	0xEA	0xA0	0xC3	0x66	0x05	0xF2	0x91	0x34	0x57	0x1D	0x7E	0xDB	0xB8
0x8?	0xC8	0xAB	0x0E	0x6D	0x27	0x44	0xE1	0x82	0x75	0x16	0xB3	0xD0	0x9A	0xF9	0x5C	0x3F
0x9?	0xD1	0xB2	0x17	0x74	0x3E	0x5D	0xF8	0x9B	0x6C	0x0F	0xAA	0xC9	0x83	0xE0	0x45	0x26
0xA?	0xFA	0x99	0x3C	0x5F	0x15	0x76	0xD3	0xB0	0x47	0x24	0x81	0xE2	0xA8	0xCB	0x6E	0x0D
0xB?	0xE3	0x80	0x25	0x46	0x0C	0x6F	0xCA	0xA9	0x5E	0x3D	0x98	0xFB	0xB1	0xD2	0x77	0x14
0xC?	0xAC	0xCF	0x6A	0x09	0x43	0x20	0x85	0xE6	0x11	0x72	0xD7	0xB4	0xFE	0x9D	0x38	0x5B
0xD?	0xB5	0xD6	0x73	0x10	0x5A	0x39	0x9C	0xFF	0x08	0x6B	0xCE	0xAD	0xE7	0x84	0x21	0x42
0xE?	0x9E	0xFD	0x58	0x3B	0x71	0x12	0xB7	0xD4	0x23	0x40	0xE5	0x86	0xCC	0xAF	0x0A	0x69
0xF?	0x87	0xE4	0x41	0x22	0x68	0x0B	0xAE	0xCD	0x3A	0x59	0xFC	0x9F	0xD5	0xB6	0x13	0x70

For instance, given the same substream bytes as before, 0x95 0x00 0x20 0x00 0x1C 0xB6, our CRC-8 calculation is as follows:

$$\begin{aligned} \text{CRC8}(\texttt{0x95} \textbf{ xor } \texttt{0x3C}) = \text{CRC8}(\texttt{0xA9}) &= \texttt{0x24} \\ \text{CRC8}(\texttt{0x00} \textbf{ xor } \texttt{0x24}) = \text{CRC8}(\texttt{0x24}) &= \texttt{0xDD} \\ \text{CRC8}(\texttt{0x20} \textbf{ xor } \texttt{0xDD}) = \text{CRC8}(\texttt{0xFD}) &= \texttt{0xB6} \\ \texttt{0xB6} \textbf{ xor } \texttt{0x00} &= \texttt{0xB6} \end{aligned}$$

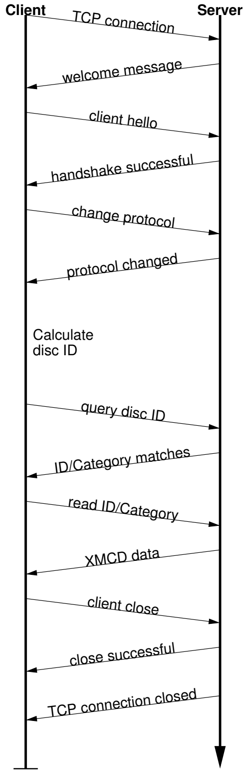
which matches our CRC-8 byte at the end of the substream.

18 FreeDB

Because compact discs do not usually contain metadata about track names, album names and so forth, that information must be retrieved from an external source. FreeDB is a service which allows users to submit CD metadata and to retrieve the metadata submitted by others. Both actions require a category and a 32-bit disc ID number, which combine to form a unique identifier for a particular CD.

18.1 Native Protocol

FreeDB's native protocol runs as a service on TCP port 8880.



- After connecting, the client and server exchange a handshake using the `hello` command. The server will not do anything without this handshake.

- Next the client changes to protocol level 6 with the `proto` command. This is necessary because only the highest protocol supports UTF-8 text encoding. Without this, any characters not in the Latin-1 set will not be sent properly.

- Once that is accomplished, the client should calculate the 32-bit disc ID from the track information.

- One then sends the 32-bit disc ID and additional disc information to the server with the `query` command to retrieve a list of matching disc IDs, genres and titles. If there are multiple matches, the user must be prompted to choose one of the matches.

- When our match is known, the client uses the `read` command to retrieve the actual XMCD data.

- Finally, the `close` command is used to sever the connection and complete the transaction.

18.1.1 the Disc ID

FreeDB uses a big-endian 32-bit disc ID to differentiate on disc from another.

Offset Seconds Digit Sum	Total Length in Seconds	Track Count
0 7	8 23	24 31

'Track Count' is self-explanatory. 'Total Length' is the total length of all the tracks, not counting the initial 2 second lead-in. 'Offset Seconds Digit Sum' is the sum of the digits of all the disc's track offsets, in seconds, and truncated to 8 bits. Remember to count the initial 2 second/150 frame lead-in when calculating offsets.

Track	Length			Offset		
Number	in M:SS	in seconds	in frames	in M:SS	in seconds	in frames
1	3:37	217	16340	0:02	2	150
2	3:23	203	15294	3:39	219	16490
3	3:37	217	16340	7:03	423	31784
4	3:20	200	15045	10:41	641	48124

In this example, 'Track Count' is **4**. 'Total Length' is $\frac{16340+15294+16340+15045}{75} = \mathbf{840}$

There are 75 frames per second, and one must remember to count fractions of seconds when calculating the total disc length.

The 'Offset Seconds Digit Sum' is calculated by looking at the 'Offset in Seconds' column. Those values are 2, 219, 423 and 641. One must take all of those digits and add them, which works out to $2+2+1+9+4+2+3+6+4+1 = \mathbf{34}$

This means our three values are 34, 840 and 4. In hexadecimal, they are 0x22, 0x0348 and 0x04. Combining them into a single value yields 0x22034804. Thus, our FreeDB disc ID is 22034804

18.1.2 Initial Greeting

From Server

```
<code>␣<host>␣CDDBP␣server␣<version>␣ready␣at␣<datetime>
```

	200	OK, reading/writing allowed
	201	OK, read-only
`<code>`	432	No connections allowed: permission denied
	433	No connections allowed: X users allowed, Y currently active
	434	No connections allowed: system load too high
`<hostname>`		the server's host name
`<version>`		the server's version
`<datetime>`		the current date and time

18.1.3 Client-Server Handshake

To Server

```
cddb␣hello␣<username>␣<hostname>␣<clientname>␣<version>
```

`<username>`	login name of user
`<hostname>`	host name of client
`<clientname>`	name of client program
`<version>`	version of client program

From Server

```
<code>␣hello␣and␣welcome␣<username>@<hostname>␣running␣<client>␣<version>
```

	200	handshake successful
`<code>`	402	already shook hands
	431	handshake unsuccessful, closing connection
`<username>`		login name of user
`<hostname>`		host name of client
`<clientname>`		name of client program
`<version>`		version of client program

18.1.4 Set Protocol Level

To Server

```
proto␣[level]
```

`[level]`	protocol level as integer (optional)

From Server

```
<code>␣CDDB␣protocol␣level:␣<current>,␣supported␣<supported>

OR

<code>␣OK,␣protocol␣version␣now:␣␣<current>
```

`<code>`	200	displaying current protocol level
	201	protocol level set
	501	illegal protocol level
	502	protocol level already at `<current>`
`<current>`		the current protocol level of this connection
`<supported>`		the maximum supported protocol level

18.1.5 Query Database

To Server

```
cddb␣query␣<disc_id>␣<track_count>␣<offset_1>␣<...>␣<offset_n>␣<seconds>
```

<disc_id>	32-bit disc ID
<track_count>	number of tracks in CD
<offset>	frame offset of each track
<seconds>	total length of CD in seconds

From Server

```
<code>␣<category>␣<disc_id>␣<disc_title>

OR

<code>␣close␣matches␣found
<category>␣<disc_id>␣<disc_title>
<category>␣<disc_id>␣<disc_title>
<...>
.

OR

<code>␣exact␣matches␣found
<category>␣<disc_id>␣<disc_title>
<category>␣<disc_id>␣<disc_title>
<...>
.
```

<code>	200	Found exact match
	211	Found inexact matches, list follows
	202	No match found
	210	Found exact matches, list follows
	403	Database entry corrupt
	409	no handshake
<category>		category string
<disc_id>		32-bit disc ID
<disc_title>		disc title string

18.1.6 Read XMCD Data

To Server

```
cddb␣read␣<category>␣<disc_id>
```

`<category>`	category string
`<disc_id>`	32-bit disc ID

From Server

```
<code>␣<category>␣<disc_id>
<XMCD_file_data>
<...>
.
```

`<code>`	210	XMCD data follows
	401	XMCD data not found
	402	server error
	403	database entry corrupt
	409	no handshake
`<category>`		category string
`<disc_id>`		32-bit disc ID

18.1.7 Close Connection

To Server

```
quit
```

From Server

```
<code>␣<hostname>␣<message>
```

`<code>`	230	Closing connection. Goodbye.
	530	error, closing connection.
`<message>`		exit message
`<hostname>`		server's host name

18.2 Web Protocol

FreeDB’s web protocol runs as a service on HTTP port 80. A web client POSTs data to a location, typically: cddb/cddb.cgi and retrieves results. This method is similar to the native protocol and the returned data is identical. However, since HTTP POST requests are stateless, there are no separate hello, proto and quit commands; these are issued along with the primary server command or are implied.

key	value
hello	<username> <hostname> <clientname> <version>
proto	<protocol>
cmd	<command>

Table 18.1: POST arguments

For example, to execute the read command on disc ID AABBCCDD in the soundtrack category, one can POST the following string:

```
cmd=read+soundtrack+aabbccdd&hello=username+hostname+audiotools+1.0&proto=6
```

18.3 XMCD

XMCD files are text files encoded either in UTF-8, ISO-8859-1 or US-ASCII. All begin with the string ‘# XMCD’. Lines are delimited by either the 0x0A character or the 0x0D 0x0A character pair. All lines must be less than 256 characters long, including delimiters. Blank lines are prohibited. Lines that begin with the ‘#’ character are comments. Curiously, the comments themselves are expected by FreeDB to contain important information such as track offsets and disc length. Fortunately, FreeDB clients can safely ignore such information unless submitting a new disc entry.

What we are interested in are the KEY=value pairs in the rest of the file.

key	value
DISCID	a comma-separated list of 32-bit disc IDs
DTITLE	an artist name and album name, separated by ‘ / ’
DYEAR	a 4 digit disc release year
DGENRE	the disc’s FreeDB category string
TITLE**X**	the track title, or the track artist name and track title, separated by ‘ / ’ **X** is an integer starting from 0
EXTD	extended data about the disc
EXTT**X**	extended data about the track **X** is an integer starting from 0
PLAYORDER	a comma-separated list of track numbers

Multiple identical keys should have their values concatenated (minus the newline delimiter), which allows a single key to have a value longer than the 256 characters line length.

19 MusicBrainz

MusicBrainz is another CD metadata retrieval service similar to FreeDB, but designed to eliminate many of FreeDB's limitations. For example, MusicBrainz has a more robust disc ID calculation mechanism, it has an easier way to disambiguate database entries in case of collision, and its XML metadata format is less prone to errors (track names with '/' characters are a particular problem for FreeDB).

However, because it is a newer service, it's common to find disc entries that are on FreeDB but do not yet have a MusicBrainz entry - whereas the converse is much more rare. Therefore, a metadata looking program would be wise to check both services if possible.

19.1 Searching Releases

This is analogous to FreeDB's search routine in which one calculates a CD's disc ID, submits it to MusicBrainz via an HTTP get query and receives information such as album name, artist name, track names and so forth as an XML file.

19.1.1 the Disc ID

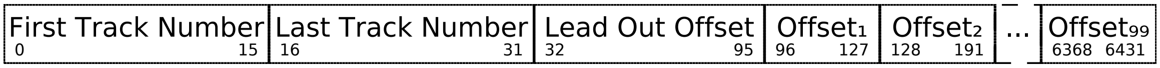
Calculating a MusicBrainz disc ID requires knowing a CD's first track number, last track number, track offsets (in CD frames) and lead out track offset (also in CD frames). For example, given the following CD:

Track	Length			Offset		
Number	in M:SS	in seconds	in frames	in M:SS	in seconds	in frames
1	3:37	217	16340	0:02	2	150
2	3:23	203	15294	3:39	219	16490
3	3:37	217	16340	7:03	423	31784
4	3:20	200	15045	10:41	641	48124

The first track number is 1, the last track number is 4, the track offsets are 150, 16490, 31784 and 48124, and the lead out track offset is 63169 (track 4's offset 48124 plus its length of 15045).

These numbers are then converted to 0-padded, big-endian hexadecimal strings with the track numbers using 2 digits and the offsets using 8 digits. In this example, the first track number becomes `01`, the last track number becomes `04`, the track offsets become `00000096`, `0000406A`, `00007C28` and `0000BBFC`, and the lead out track offset becomes `0000F6C1`.

These individual strings are then combined into a single 804 byte string:



Excess track offsets are treated as having an offset value of 0, or a string value of `00000000`. Our string starts with `01040000F6C1000000960000406A00007C280000BBFC` and is padded with an additional 760 '`0`' characters which I'll omit for brevity.

That string is then passed through the SHA-1 hashing algorithm[1] which results in a 20 byte hash value. Remember to use the binary hash value, not its 40 byte ASCII hexadecimal one.

In our example, this yields the hash: `0xDA3D930462773DD57BBE43B535AD6A457138F079`

The resulting hash value is then encoded to a 28 byte Base64[2] string. However, unlike standard Base64, MusicBrainz's disc ID replaces the characters '`=`', '`+`' and '`/`' with '`-`', '`.`' and '`_`' respectively to make the value better suited to HTTP requests. So to complete our example, the hash value becomes a disc ID of `2j2TBGJ3PdV7vkO1Na1qRXE48Hk-`

[1] This is described in RFC3174

[2] This is described in RFC3548 and RFC4648

19.1.2 Server Query

MusicBrainz runs as a service on HTTP port 80. To retrieve Release information, one can make a GET request to `/ws/1/release` using the following fields:

key	value
`type`	`xml`
`discid`	`<disc ID string>`

For example, to retrieve the Release data for disc ID `2jmj7l5rSw0yVb_vlWAYkK_YBwk-` one sends the GET query:

```
type=xml&discid=2jmj7l5rSw0yVb_vlWAYkK_YBwk-
```

Whether the Release is found in the MusicBrainz database or not, an XML file will always be generated.

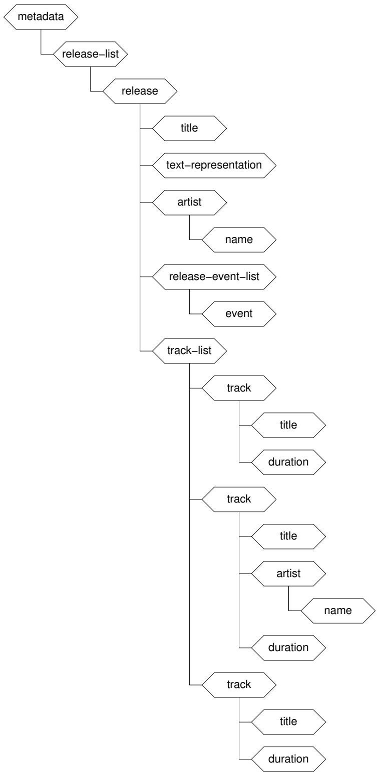
19.1.3 Release XML

All XML files returned by a MusicBrainz query consist of a `<metadata>` tag container. When making a Release query, it contains a `<release-list>` which is itself a container for zero or more `<release>` tags, depending on how many Release entries match the submitted disc ID.

The `<release>` tag typically contains a `<title>` which is the album's name, an `<artist>` tag which is the album's primary artist, a `<release-event-list>` tag containing information such as the album's release date and catalog number, and finally a `<track-list>` which contains all the track data.

The `<track>` tags are always listed in order of their appearance in the album. Each contains a `<title>` which is the track's name, a `<duration>` which is the track's length in milliseconds, and optionally an `<artist>` tag which is information about a track-specific artist, for instances where the track's artist differs from the album's artist.

In addition, the `<release>` , `<artist>` , and `<track>` tags all contain an 'id' attribute with 32 hex digits in the format '12345678-9abc-def1-2345-6789abcdef1234'. These uniquely identify the Release, Artist and Track information in the MusicBrainz database and can be used for direct lookups.



19.2 MusicBrainz XML

The following is the complete specification for MusicBrainz XML output in RELAX NG Compact syntax from http://bugs.musicbrainz.org/browser/mmd-schema/trunk/schema and converted to compact syntax for better readability.

Schema Start

```
default namespace id3034801 = "http://musicbrainz.org/ns/mmd-1.0#"

namespace local = ""

namespace inh = inherit

start = def_metadata-element

def_metadata-element =
   element metadata
   {
      attribute generator { xsd:anyURI }?,
      attribute created { xsd:dateTime }?,
      def_artist-element?,
      def_release-element?,
      def_release-group-element?,
      def_track-element?,
      def_label-element?,
      def_artist-list?,
      def_release-list?,
      def_release-group-list?,
      def_track-list?,
      def_label-list?,
      def_metadata-element_extension
   }

def_artist-element =
   element artist
   {
      attribute id { xsd:anyURI }?,
      attribute type { xsd:anyURI }?,
      def_artist-attribute_extension,
      element name { text }?,
      element sort-name { text }?,
      element disambiguation { text }?,
      element life-span
      {
         attribute begin { def_incomplete-date }?,
         attribute end { def_incomplete-date }?
      }?,
      def_alias-list?,
      def_release-list?,
      def_release-group-list?,
      def_relation-list*,
      def_tag-list?,
      def_user-tag-list?,
      def_rating?,
      def_user-rating?,
      def_artist-element_extension
   }
```

```
def_release-element =
   element release
   {
      attribute id { xsd:anyURI }?,
      attribute type { def_URI-list }?,
      def_release-attribute_extension,
      element title { text }?,
      element text-representation
      {
         attribute language { def_iso-639 }?,
         attribute script { def_iso-15924 }?
      }?,
      element asin { xsd:string { pattern = "[A-Z0-9]{10}" } }?,
      def_artist-element?,
      def_release-group-element?,
      def_release-event-list?,
      def_disc-list?,
      def_puid-list?,
      def_track-list?,
      def_relation-list*,
      def_tag-list?,
      def_user-tag-list?,
      def_rating?,
      def_user-rating?,
      def_release-element_extension
   }

def_release-group-element =
   element release-group
   {
      attribute id { xsd:anyURI }?,
      attribute type { def_URI-list }?,
      def_release-group-attribute_extension,
      element title { text }?,
      def_artist-element?,
      def_release-list?,
      def_release-group-element_extension
   }

def_track-element =
   element track
   {
      attribute id { xsd:anyURI }?,
      def_track-attribute_extension,
      element title { text }?,
      element duration { xsd:nonNegativeInteger }?,
      element isrc-list { element isrc { attribute id { def_isrc } }* }?,
      def_artist-element?,
      def_release-list?,
      def_puid-list?,
      def_relation-list*,
      def_tag-list?,
      def_user-tag-list?,
      def_rating?,
      def_user-rating?,
      def_track-element_extension
   }
```

```
def_label-element =
   element label
   {
      attribute id { xsd:anyURI }?,
      attribute type { xsd:anyURI }?,
      def_label-attribute_extension,
      element name { text }?,
      element sort-name { text }?,
      element label-code { xsd:nonNegativeInteger }?,
      element disambiguation { text }?,
      element country { def_iso-3166 }?,
      element life-span
      {
         attribute begin { def_incomplete-date }?,
         attribute end { def_incomplete-date }?
      }?,
      def_alias-list?,
      def_release-list?,
      def_release-group-list?,
      def_relation-list*,
      def_tag-list?,
      def_user-tag-list?,
      def_rating?,
      def_user-rating?,
      def_label-element_extension
   }

def_relation-element =
   element relation
   {
      attribute type { xsd:anyURI },
      attribute target { xsd:anyURI },
      attribute direction { def_direction }?,
      attribute attributes { def_URI-list }?,
      attribute begin { def_incomplete-date }?,
      attribute end { def_incomplete-date }?,
      (
         def_artist-element
       | def_release-element
       | def_track-element
       | def_relation-element_extension
      )?
   }

def_alias =
   element alias
   {
      attribute type { xsd:anyURI }?,
      attribute script { def_iso-15924 }?,
      text
   }

def_tag = element tag { attribute count { xsd:nonNegativeInteger }?, text }

def_user-tag = element user-tag { text }

def_rating =
   element rating
   {
      attribute votes-count { xsd:nonNegativeInteger }?,
      xsd:float
   }

def_user-rating = element user-rating { xsd:nonNegativeInteger }

def_metadata-element_extension = def_extension_element?

def_artist-element_extension = def_extension_element*

def_release-element_extension = def_extension_element*

def_release-group-element_extension = def_extension_element*

def_track-element_extension = def_extension_element*

def_label-element_extension = def_extension_element*

def_relation-element_extension = def_extension_element

def_artist-attribute_extension = def_extension_attribute*

def_release-attribute_extension = def_extension_attribute*

def_release-group-attribute_extension = def_extension_attribute*

def_track-attribute_extension = def_extension_attribute*

def_label-attribute_extension = def_extension_attribute*

def_extension_element =
   element * - (id3034801:* | local:*)
   {
      ( attribute * { text } | text | def_anything )*
   }

def_extension_attribute = attribute * - (id3034801:* | local:*) { text }

def_anything =
   element * - local:* { ( attribute * { text } | text | def_anything )* }

def_artist-list =
   element artist-list { def_list-attributes, def_artist-element* }

def_release-list =
   element release-list { def_list-attributes, def_release-element* }

def_release-group-list =
   element release-group-list
   {
      def_list-attributes,
      def_release-group-element*
   }

def_alias-list = element alias-list { def_list-attributes, def_alias* }

def_track-list = element track-list { def_list-attributes, def_track-element* }

def_label-list = element label-list { def_list-attributes, def_label-element* }
```

```
def_release-event-list =
   element release-event-list
   {
      def_list-attributes,
      element event
      {
         attribute date { def_incomplete-date },
         attribute country { def_iso-3166 }?,
         attribute catalog-number { text }?,
         attribute barcode { text }?,
         attribute format { xsd:anyURI }?,
         def_label-element?
      }*
   }

def_disc-list =
   element disc-list
   {
      def_list-attributes,
      element disc
      {
         attribute id { xsd:string { pattern = "[a-zA-Z0-9._]{27}-" } },
         attribute sectors { xsd:nonNegativeInteger }?
      }*
   }

def_puid-list =
   element puid-list
   {
      def_list-attributes,
      element puid { attribute id { def_uuid } }*
   }

def_relation-list =
   element relation-list
   {
      attribute target-type { xsd:anyURI },
      def_list-attributes,
      def_relation-element*
   }

def_tag-list = element tag-list { def_list-attributes, def_tag* }

def_user-tag-list =
   element user-tag-list { def_list-attributes, def_user-tag* }

def_list-attributes =
   attribute count { xsd:nonNegativeInteger }?,
   attribute offset { xsd:nonNegativeInteger }?

def_URI-list = list { xsd:anyURI+ }

def_incomplete-date =
   xsd:string { pattern = "[0-9]{4}(-[0-9]{2})?(-[0-9]{2})?" }

def_iso-3166 = xsd:string { pattern = "[A-Z]{2}" }

def_iso-639 = xsd:string { pattern = "[A-Z]{3}" }
```

```
def_iso-15924 = xsd:string { pattern = "[A-Z][a-z]{3}" }

def_isrc = xsd:string { pattern = "[A-Z]{2}[A-Z0-9]{3}[0-9]{2}[0-9]{5}" }

def_uuid = xsd:string { pattern = "[0-9a-f]{8}(-[0-9a-f]{4}){3}-[0-9a-f]{12}" }

def_direction = "both" | "forward" | "backward"
```

Schema End

20 ReplayGain

The ReplayGain standard is designed to address the problem of highly variable music loudness. For example, let's assume we have two audio tracks, A and B, and that track B is much louder than A. If played in sequence, the listener will have to scramble for the volume control once B starts in order to have a comfortable experience. ReplayGain solves this problem by calculating the overall loudness of a track as a delta (some positive or negative number of decibels, in relation to a reference loudness value). This delta is then applied during playback, which has the same effect as turning the volume up or down so that the user doesn't have to.

ReplayGain requires four floating-point values which are typically stored as metadata in each audio track: 'track gain', a positive or negative number of decibels representing the loudness delta of this particular track, 'track peak', the highest sample value of this particular track from a range of 0.0 to 1.0, 'album gain', a positive or negative number of decibels representing the loudness delta of the track's entire album and 'album peak', the highest sample value of the track's entire album from a range of 0.0 to 1.0.

20.1 Applying ReplayGain

The user will be expected to choose whether to apply 'album gain' or 'track gain' during playback. When listening to audio on an album-by-album basis, album gain keeps quiet tracks quiet and loud tracks loud within the context of that album. When listening to audio on a track-by-track basis, perhaps as a randomly shuffled set, track gain keeps them all to roughly the same loudness. So from an implementation perspective, a program only needs to apply the given gain and peak value to the stream being played back. Applying the gain value to each input PCM sample is quite simple:

$$\text{Output}_i = \text{Input}_i \times 10^{\frac{\text{gain}}{20}} \tag{20.1}$$

For example, if the gain is -2.19, each sample should be multiplied by $10^{\frac{-2.19}{20}}$ or about 0.777.

If the gain is negative, the PCM stream gets quieter than it was originally. If the gain is positive, the PCM stream gets louder. However, increasing the value of each sample may cause a problem if doing so sends any samples beyond the maximum value the stream can hold. For example, if the gain indicates we should be multiplying each sample by 1.28 and we encounter a 16-bit input sample with a value of 32000, the resulting output sample of 34560 is outside of the stream's 16-bit signed range (-32678 to 32767). That will result in 'clipping' the audio peaks, which doesn't sound good.

Preventing this is what ReplayGain's peak value is for; it's the highest PCM value in the stream and no multiplier should push that value beyond 1.0. Thus, if the peak value of a stream is 0.9765625, no ReplayGain value should generate a multiplier higher than 1.024 ($0.9765625 \times 1.024 = 1.0$).

20.2 Calculating ReplayGain

As explained earlier, ReplayGain requires a peak and gain value which are split into 'track' and 'album' varieties for a total of four. The 'track' values require the PCM data for the particular track we're generating data for. The 'album' values require the PCM data for the entire album, concatenated together into a single stream.

Determining the peak value is very straightforward. We simply convert each sample's value to the range of 0.0 to 1.0 and find the highest value which occurs in the stream. For signed samples, the conversion process is also simple:

$$\text{Output}_i = \frac{|\text{Input}_i|}{2^{\text{bits per sample}-1}} \tag{20.2}$$

Determining the gain value is a more complicated process. It involves running the input stream through an equal loudness filter, breaking that stream into 50 millisecond long blocks, and then determining a final value based on the value of those blocks.

20.2.1 the Equal Loudness Filter

Because people don't perceive all frequencies of sounds as having equal loudness, ReplayGain runs audio through a filter which emphasizes ones we hear as loud and deemphasizes ones we hear as quiet. This equal loudness filtering is actually comprised of two separate filters: Yule and Butter (these are Infinite Impulse Response filters named after their creators). Each works on a similar principle.

The basic premise is that each output sample is derived from multiplying 'order' number of previous input samples by certain values (which depend on the filter) *and* 'order' number of previous output samples by a different set of values (also depending on the filter) and then combining the results. This filter is applied independently to each channel. In purely mathematical terms, it looks like this:

$$\text{Output}_i = \left(\sum_{j=i-order}^{i} \text{Input}_j \times \text{Input Filter}_j\right) - \left(\sum_{k=i-order}^{i-1} \text{Output}_k \times \text{Output Filter}_k\right) \tag{20.3}$$

'Input Filter' and 'Output Filter' are lists of predefined values. 'Order' refers to the size of those lists. When filtering at the start of the stream, treat any samples before the beginning as 0.

a filtering example

Let's assume we have a 44100Hz stream and our previous input and output samples are as follows:

sample	Input_i	Yule_i	Butter_i
89	-33	-14.90	
90	-32	-14.93	
91	-35	-14.65	
92	-32	-14.46	
93	-30	-14.15	
94	-32	-13.58	
95	-33	-13.18	
96	-30	-13.16	
97	-30	-13.12	0.41
98	-30	-12.89	0.61
99	-32	-12.81	0.66

If the value of sample 100 from the input stream is -30, here's how we calculate output sample 100:

sample	Input_i		Yule Input Filter_i		result	Yule_i		Yule Output Filter_i		result
90	-32	×	-0.00187763777362	=	0.06	-14.93	×	0.13149317958807999	=	-1.96
91	-35	×	0.0067461368224699999	=	-0.24	-14.65	×	-0.75104302451432003	=	11.00
92	-32	×	-0.0024087905158400001	=	0.08	-14.46	×	2.1961168489077401	=	-31.76
93	-30	×	0.016248649629749999	=	-0.49	-14.15	×	-4.3947099607955904	=	62.19
94	-32	×	-0.025963385129149998	=	0.83	-13.58	×	6.8540154093699801	=	-93.08
95	-33	×	0.022452932533390001	=	-0.74	-13.18	×	-8.8149868137015499	=	116.18
96	-30	×	-0.0083499090493599996	=	0.25	-13.16	×	9.4769360780128	=	-124.72
97	-30	×	-0.0085116564546900003	=	0.26	-13.12	×	-8.5475152747187408	=	112.14
98	-30	×	-0.0084870937985100006	=	0.25	-12.89	×	6.3631777756614802	=	-82.02
99	-32	×	-0.029110078089480001	=	0.93	-12.81	×	-3.4784594855007098	=	44.56
100	**-30**	×	0.054186564064300002	=	-1.63					
			input values sum	=	-0.44			output values sum	=	12.53

Therefore, $\text{Yule}_{100} = -0.44 - 12.53 = -12.97$

We're not quite done yet. Remember, ReplayGain's equal loudness filter requires both a Yule *and* Butter filter, in that order. Notice how Butter's input samples are Yule's output samples. Thus, our next input sample to the Butter filter is -12.97. Calculating sample 100 is now a similar process:

sample	Yule_i		Butter Input Filter_i		result	Butter_i		Butter Output Filter_i		result
98	-12.89	×	0.98500175787241995	=	-12.70	0.61	×	0.97022847566350001	=	0.59
99	-12.81	×	-1.9700035157448399	=	25.24	0.66	×	-1.96977855582618	=	-1.30
100	**-12.97**	×	0.98500175787241995	=	-12.78					
			input values sum	=	-0.24			output values sum	=	-0.71

Therefore, $\text{Butter}_{100} = -0.24 - -0.71 = 0.47$, which is the next sample from the equal loudness filter.

20.2.2 RMS Energy Blocks

The next step is to take our stream of filtered samples and convert them to a list of blocks, each 1/20th of a second long. For example, a 44100Hz stream is sliced into blocks containing 2205 PCM frames each.

We then figure out the total energy value of each block by taking the Root Mean Square of the block's samples and converting to decibels, hence the name RMS.

$$\text{Block dB}_i = 10 \times \log_{10}\left(\frac{\left(\frac{\sum_{x=0}^{\text{Block Length}-1} \text{Left Sample}_x{}^2}{\text{Block Length}}\right) + \left(\frac{\sum_{y=0}^{\text{Block Length}-1} \text{Right Sample}_y{}^2}{\text{Block Length}}\right)}{2} + 10^{-10}\right) \tag{20.4}$$

For mono streams, use the same value for both the left and right samples (this will cause the addition and dividing by 2 to cancel each other out). As a partial example involving 2205 PCM frames:

Sample	Left Value	Left Value^2	Right Value	Right Value^2
998	115	13225	-43	1849
999	111	12321	-38	1444
1000	107	11449	-36	1296
...	...		...	
	Left Value^2 sum = 7106715		Right Value^2 sum = 11642400	

$$\frac{(\frac{7106715}{2205}) + (\frac{11642400}{2205})}{2} = 4251 \tag{20.5}$$

$$10 \times \log_{10}(4251 + 10^{-10}) = 36.28 \tag{20.6}$$

Thus, the decibel value of this block is 36.28.

20.2.3 Statistical Processing and Calibration

At this point, we've converted our stream of input samples into a list of RMS energy blocks. We now pick the 95th percentile value as the audio stream's representative value. That means we first sort them from lowest to highest, then pick the one at the 95% position. For example, if we have a total of 2400 decibel blocks (from a 2 minute song), the value of block 2280 is our representative.

Finally, we take the difference between a reference value of pink noise and our representative value for the final gain value. The reference pink noise value is typically 64.82 dB. Therefore, if our representative value is 67.01 dB, the resulting gain value is -2.19 dB $(64.82 - 67.01 = -2.19)$.

Appendices

A References

- Wave File Format Specifications
  `http://www-mmsp.ece.mcgill.ca/Documents/AudioFormats/WAVE/WAVE.html`

- Audio File Format Specifications
  `http://www-mmsp.ece.mcgill.ca/Documents/AudioFormats/AIFF/AIFF.html`

- AU Audio File Format
  `http://www.opengroup.org/public/pubs/external/auformat.html`

- Shorten Research Paper
  `ftp://svr-ftp.eng.cam.ac.uk/pub/reports/robinson_tr156.ps.Z`

- FLAC Format Specification
  `http://flac.sourceforge.net/format.html`

- APEv2 Specification
  `http://wiki.hydrogenaudio.org/index.php?title=APEv2_specification`

- WavPack 4.0 File / Block Format
  `http://www.wavpack.com/file_format.txt`

- MPEG Audio Compression Basics
  `http://www.datavoyage.com/mpgscript/mpeghdr.htm`

- What is ID3v1
  `http://www.id3.org/ID3v1`

- The ID3v2 Documents
  `http://www.id3.org/Developer_Information`

- The Ogg File Format
  `http://en.wikipedia.org/wiki/Ogg#File_format`

- Vorbis I Specification
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- Proposals for Extending Ogg Vorbis Comments
  `http://www.reallylongword.org/articles/vorbiscomment/`

- Speex Documentation
  http://www.speex.org/docs/
- Musepack Stream Version 7 Format Specification
  http://trac.musepack.net/trac/wiki/SV7Specification
- Parsing and Writing QuickTime Files in Java
  http://www.onjava.com/pub/a/onjava/2003/02/19/qt_file_format.html
- ALAC decoder
  http://craz.net/programs/itunes/alac.html
- ISO 14496-1 Media Format
  http://xhelmboyx.tripod.com/formats/mp4-layout.txt
- FreeDB Information
  http://www.freedb.org/en/download__miscellaneous.11.html
- ReplayGain
  http://replaygain.hydrogenaudio.org

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