Ogg/Vorbis

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Voorwoord

Dit document is geschreven in het kader van het onderzoek van Lucien Immink wat hij schreef in het vierde jaar van de opleiding Media Technologie. Van de student wordt verwacht een literair onderzoek te doen naar een relevant onderdeel van de opleiding, dit kan iets zijn wat de student is tegen gekomen tijdens de lessen, maar mag ook iets zijn wat de student interesseert. De totale duur van het onderzoek bedraagt 80 uur, wat gelijk staat aan 2 studiepunten.

Ik heb voor dit specifieke onderzoek gekozen omdat het onderwerp me al enkele jaren meer dan boeit. Hoe ogg/vorbis precies in elkaar zat wist voor het schrijven van dit stuk nog niet. Hoe de verschillende onderdelen het formaat maakte en hoe het formaat zich vergeleek met andere formaten was me ook nog niet geheel duidelijk. Het verslag zal hier uitsluitsel over geven.

Samenvatting

Inhoudsopgave

1 Inleiding

Dit onderzoek is verricht voor de opleiding MediaTechnologie van de Hogeschool van Utrecht te Amersfoort. Gesteld is dat elke student een literair onderzoek moet verrichten binnen het vakgebied van de opleiding. Met onderzoek wordt dan bedoeld: het verzamelen, bewerken en analyseren van gegevens om meer te weten te komen. Onderzoek is het combineren van gegevens afkomstig uit verschillende bronnen om tot nieuwe inzichten te komen.
Binnen dit kader is er voor dit specifieke onderzoek gekozen om het volgende te onderzoeken:

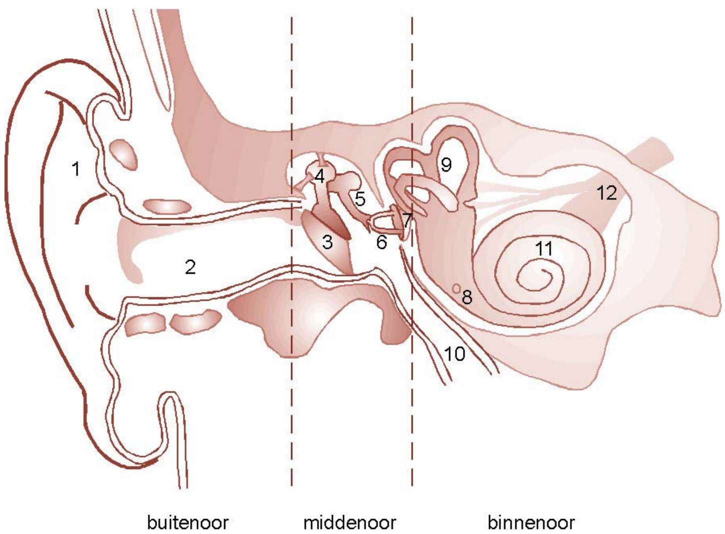
2 Het menselijk oor

Om goed te begrijpen hoe een computer 3D geluid kan produceren is het noodzakelijk het menselijk oor te kennen. In het volgende hoofdstuk wordt uitgelegd hoe het menselijk oor in elkaar zit. Dit hoofdstuk zal slechts een overzicht bieden van het oor, aangezien het gehoor al reeds behandeld is binnen de opleiding. Voor geïnteresseerden staat er achter in een literatuurlijst met links naar websites die geheel over het gehoor gaan.

2.1 Overzicht

Het is gebruikelijk het oor onder te verdelen in drie stukken, het buitenoor, het middenoor en het binnenoor, zoals aangegeven in fig. 2.1. tezamen ook wel genoemd het *perifere gehoor*.

Het *buitenoor* omvat de oorschelp en de gehoorgang, die loopt tot aan het trommelvlies. De oorschelp helpt bij het richting horen. De gehoorgang functioneert als een resonator die frequenties tussen de 1 en de 4 kHz versterkt; dit is precies het frequentiegebied dat voor spraak van belang is.



Figuur 2.1

Het middenoor bevindt zich tussen het trommelvlies aan de buitenkant en het ovale venster meer hoof dinwaarts. De luchtdrukverschillen duwen het trommelvlies beurtelings naar binnen en zuigen het naar buiten. Via drie minuscule botjes, die - naar hun uiterlijk - hamer, aambeeld en stijgbeugel heten, wordt de beweging van het trommelvlies doorgegeven aan het ovale venster. Omdat het ovale venster een veel kleiner oppervlak heeft dan het trommelvlies en omdat de botjes als een hefboom werken, worden de zwakke bewegingen van het trommelvlies mechanisch versterkt. De botjes kunnen in hun bewegingen geremd worden door kleine spiertjes die reflexmatig worden aangespannen wanneer het oor getroffen wordt door heel luide, lage trillingen. Na een rockconcert blijven we enige tijd hardhorend omdat deze zelfbeschermingreflex zich niet meteen ontspant. Helaas

komt de reflex te langzaam op gang om ook gehoorbeschadiging als gevolg van explosies te voorkomen.
Het binnenoor is voor de waarneming van spraak en muziek het belangrijkste deel van het perifere gehoor. Hier worden de fysieke geluidstrillingen omgezet in elektrische trillingen voor verdere verwerking in het centrale gehoor (d.w.z. in de hersenen).

2.2 Samenvatting

Dit onderzoek is gericht op de vraag of de akoestiek van de ene ruimte over te brengen is in een willekeurige andere ruimte. Het is dan van belang om te begrijpen hoe de ontvanger het geluid hoort. Daarom is het van belang een inzicht te hebben in de werking van het menselijk oor. Het buitenoor omvat de oorschelp en de gehoorgang. De gehoorgang functioneert als een resonator die frequenties tussen de 1 en de 4 kHz versterkt. Het middenoor bevindt zich tussen het trommelvlies aan de buitenkant en het ovale venster meer hoofd inwaarts. Het binnenoor is voor de waarneming van spraak en muziek het belangrijkste deel van het perifere gehoor.

3 Ogg

Ogg codecs use octet vectors of raw, compressed data (*packets*). These compressed packets do not have any high-level structure or boundary information; strung together, they appear to be streams of random bytes with no landmarks.

Raw packets may be used directly by transport mechanisms that provide their own framing and packet-separation mechanisms (such as UDP datagrams). For stream based storage (such as files) and transport (such as TCP streams or pipes), Vorbis and other future Ogg codecs use the Ogg bitstream format to provide framing/sync, sync recapture after error, landmarks during seeking, and enough information to properly separate data back into packets at the original packet boundaries without relying on decoding to find packet boundaries.

3.1 Logical and physical bitstreams

Raw packets are grouped and encoded into contiguous pages of structured bitstream data called *logical bitstreams*. A logical bitstream consists of pages, in order, belonging to a single codec instance. Each page is a self contained entity (although it is possible that a packet may be split and encoded across one or more pages); that is, the page decode mechanism is designed to recognize, verify and handle single pages at a time from the overall bitstream.
Multiple logical bitstreams can be combined (with restrictions) into a single *physical bitstream*. A physical bitstream consists of multiple logical bitstreams multiplexed at the page level and may include a 'meta-header' at the beginning of the multiplexed logical stream that serves as identification magic. Whole pages are taken in order from multiple logical bitstreams and combined into a single physical stream of pages. The decoder reconstructs the original logical bitstreams from the physical bitstream by taking the pages in order from the physical bitstream and redirecting them into the appropriate logical decoding entity. The simplest physical bitstream is a single, unmultiplexed logical bitstream with no meta-header; this is referred to as a 'degenerate stream'.
Ogg Logical Bitstream Framing discusses the page format of an Ogg bitstream, the packet coding process and logical bitstreams in detail. The remainder of this document specifies requirements for constructing finished, physical Ogg bitstreams.

3.2 Mapping Restrictions

Logical bitstreams may not be mapped/multiplexed into physical bitstreams without restriction. Here we discuss design restrictions on Ogg physical bitstreams in general, mostly to introduce design rationale. Each 'media' format defines its own (generally more restrictive) mapping. An 'Ogg Vorbis Audio Bitstream', for example, has a specific physical bitstream structure. An 'Ogg A/V' bitstream (not currently specified) will also mandate a specific, restricted physical bitstream format.

3.2.1 additional end-to-end structure

The framing specification defines 'beginning of stream' and 'end of stream' page markers via a header flag (it is possible for a stream to consist of a single page). A stream always consists of an integer number of pages, an easy requirement given the variable size nature of pages.
In addition to the header flag marking the first and last pages of a logical bitstream, the first page of an Ogg bitstream obeys additional restrictions. Each individual media mapping specifies its own implementation details regarding these restrictions.

The first page of a logical Ogg bitstream consists of a single, small 'initial header' packet that includes sufficient information to identify the exact CODEC type and media requirements of the logical bitstream. The intent of this restriction is to simplify identifying the bitstream type and content; for a given media type (or across all Ogg media types) we can know that we only need a small, fixed amount of data to uniquely identify the bitstream type.
As an example, Ogg Vorbis places the name and revision of the Vorbis CODEC, the audio rate and the audio quality into this initial header, thus simplifying vastly the certain identification of an Ogg Vorbis audio bitstream.

3.2.2 sequential multiplexing (chaining)

The simplest form of logical bitstream multiplexing is concatenation (*chaining*). Complete logical bitstreams are strung one-after-another in order. The bitstreams do not overlap; the final page of a given logical bitstream is immediately followed by the initial page of the next. Chaining is the only logical->physical mapping allowed by Ogg Vorbis.
Each chained logical bitstream must have a unique serial number within the scope of the physical bitstream.

3.2.3 concurrent multiplexing (grouping)

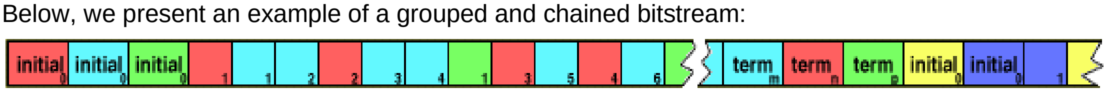
Logical bitstreams may also be multiplexed 'in parallel' (*grouped*). An example of grouping would be to allow streaming of separate audio and video streams, using different codecs and different logical bitstreams, in the same physical bitstream. Whole pages from multiple logical bitstreams are mixed together.
The initial pages of each logical bitstream must appear first; the media mapping specifies the order of the initial pages. For example, Ogg A/V will eventually specify an Ogg video bitstream with audio. The mapping may specify that the physical bitstream must begin with the initial page of a logical video bitstream, followed by the initial page of an audio stream. Unlike initial pages, terminal pages for the logical bitstreams need not all occur contiguously (although a specific media mapping may require this; it is not mandated by the generic Ogg stream spec). Terminal pages may be 'nil' pages, that is, pages containing no content but simply a page header with position information and the 'last page of bitstream' flag set in the page header.
Each grouped bitstream must have a unique serial number within the scope of the physical bitstream.

3.2.4 sequential and concurrent multiplexing

Groups of concurrently multiplexed bitstreams may be chained consecutively. Such a physical bitstream obeys all the rules of both grouped and chained multiplexed streams; the groups, when unchained , must stand on their own as a valid concurrently multiplexed bitstream.

3.2.5 multiplexing example

Below, we present an example of a grouped and chained bitstream:



In this example, we see pages from five total logical bitstreams multiplexed into a physical bitstream. Note the following characteristics:

1. Grouped bitstreams begin together; all of the initial pages must appear before any data pages. When concurrently multiplexed groups are chained, the new group does not begin until all the bitstreams in the previous group have terminated.
2. The pages of concurrently multiplexed bitstreams need not conform to a regular order; the only requirement is that page n of a logical bitstream follow page n-1 in the physical bitstream. There are no restrictions on intervening pages belonging to other logical bitstreams. (Tying page appearance to bitrate demands is one logical strategy, ie, the page appears at the chronological point where decode requires more information).

4 Vorbis

4.1 Application

Vorbis is a general purpose perceptual audio CODEC intended to allow maximum encoder flexibility, thus allowing it to scale competitively over an exceptionally wide range of bitrates. At the high quality/bitrate end of the scale (CD or DAT rate stereo, 16/24 bits), it is in the same league as MPEG-2 and MPC. Similarly, the 1.0 encoder can encode high-quality CD and DAT rate stereo at below 48kpbs without resampling to a lower rate. Vorbis is also intended for lower and higher sample rates (from 8kHz telephony to 192kHz digital masters) and a range of channel representations (monaural, polyphonic, stereo, quadraphonic, 5.1, ambisonic, or up to 255 discrete channels).

4.2 Classification

Vorbis I is a forward-adaptive monolithic transform CODEC based on the Modified Discrete Cosine Transform. The codec is structured to allow addition of a hybrid wavelet filterbank in Vorbis II to offer better transient response and reproduction using a transform better suited to localized time events.

4.3 Assumptions

The Vorbis CODEC design assumes a complex, psychoacoustically-aware encoder and simple, low-complexity decoder. Vorbis decode is computationally simpler than mp3, although it does require more working memory as Vorbis has no static probability model; the vector codebooks used in the first stage of decoding from the bitstream are packed, in their entirety, into the Vorbis bitstream headers. In packed form, these codebooks occupy only a few kilobytes; the extent to which they are pre-decoded into a cache is the dominant factor in decoder memory usage.
Vorbis provides none of its own framing, synchronization or protection against errors; it is solely a method of accepting input audio, dividing it into individual frames and compressing these frames into raw, unformatted 'packets'. The decoder then accepts these raw packets in sequence, decodes them, synthesizes audio frames from them, and reassembles the frames into a facsimile of the original audio stream. Vorbis is a free-form VBR codec and packets have no minimum size, maximum size, or fixed/expected size. Packets are designed that they may be truncated (or padded) and remain decodable; this is not to be considered an error condition and is used extensively in bitrate management in peeling. Both the transport mechanism and decoder must allow that a packet may be any size, or end before or after packet decode expects.
Vorbis packets are thus intended to be used with a transport mechanism that provides free-form framing, sync, positioning and error correction in accordance with these design assumptions, such as Ogg (for file transport) or RTP (for network multicast). For purposes of a few examples in this document, we will assume that Vorbis is to be embedded in an Ogg stream specifically, although this is by no means a requirement or fundamental assumption in the Vorbis design.
The specifications for embedding Vorbis into an Ogg transport stream is in a separate document.

4.4 Codec Setup and Probability Model

Vorbis's heritage is as a research CODEC and its current design reflects a desire to allow multiple decades of continuous encoder improvement before running out of room within the codec specification. For these reasons, configurable aspects codec setup intentionally lean toward the extreme of forward adaptive.

The single most controversial design decision in Vorbis [and the most unusual for a Vorbis developer to keep in mind] is that the entire probability model of the codec, the Huffman and VQ codebooks, is packed into the bitstream header along with extensive CODEC setup parameters (often several hundred fields). This makes it impossible, as it would be with MPEG audio layers, to embed a simple frame type flag in each audio packet, or begin decode at any frame in the stream without having previously fetched the codec setup header. [Note: Vorbis *can* initiate decode at any arbitrary packet within a bitstream so long as the codec has been initialized/setup with the setup headers].
Thus, Vorbis headers are both required for decode to begin and relatively large as bitstream headers go. The header size is unbounded, although for streaming a rule-of-thumb of 4kB or less is recommended (and Xiph.Org's Vorbis encoder follows this suggestion).
Our own design work indicates the the primary liability of the required header is in mindshare; it is an unusual design and thus causes some amount of complaint among engineers as this runs against current design trends (and also points out limitations in some existing software/interface designs, such as Windows' ACM codec framework). However, we find that it does not fundamentally limit Vorbis's suitable application space.

4.5 Format Specification

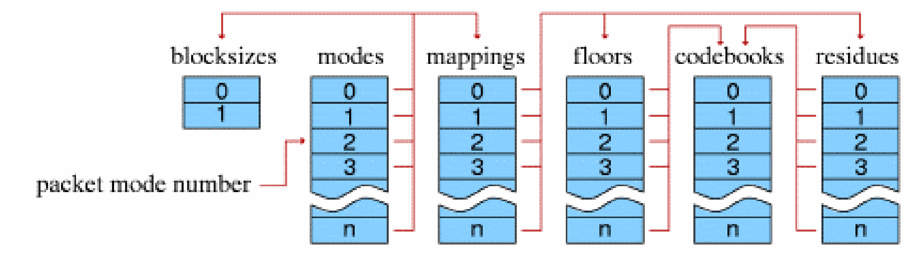
The Vorbis format is well-defined by its decode specification; any encoder that produces packets that are correctly decoded by the reference Vorbis decoder described below may be considered a proper Vorbis encoder. A decoder must faithfully and completely implement the specification defined below [except where noted] to be considered a proper Vorbis decoder.

4.6 Hardware Profile

Although Vorbis decode is computationally simple, it may still run into specific limitations of an embedded design. For this reason, embedded designs are allowed to deviate in limited ways from the 'full' decode specification yet still be certified compliant. These optional omissions are labelled in the spec where relevant.

5 Decoder Configuration

Decoder setup consists of configuration of multiple, self-contained component abstractions that perform specific functions in the decode pipeline. Each different component instance of a specific type is semantically interchangeable; decoder configuration consists both of internal component configuration, as well as arrangement of specific instances into a decode pipeline. Componentry arrangement is roughly as follows:



5.1 Global Config

Global codec configuration consists of a few audio related fields (sample rate, channels), Vorbis version (always '0' in Vorbis I), bitrate hints, and the lists of component instances. All other configuration is in the context of specific components.

5.2 Mode

Each Vorbis frame is coded according to a master 'mode'. A bitstream may use one or many modes. The mode mechanism is used to encode a frame according to one of multiple possible methods with the intention of choosing a method best suited to that frame. Different modes are, e.g. how frame size is changed from frame to frame. The mode number of a frame serves as a top level configuration switch for all other specific aspects of frame decode.
A 'mode' configuration consists of a frame size setting, window type (always 0, the Vorbis window, in Vorbis I), transform type (always type 0, the MDCT, in Vorbis I) and a mapping number. The mapping number specifies which mapping configuration instance to use for low-level packet decode and synthesis.

5.3 Mapping

A mapping contains a channel coupling description and a list of 'submaps' that bundle sets of channel vectors together for grouped encoding and decoding. These submaps are not references to external components; the submap list is internal and specific to a mapping.
A 'submap' is a configuration/grouping that applies to a subset of floor and residue vectors within a mapping. The submap functions as a last layer of indirection such that specific special floor or residue settings can be applied not only to all the vectors in a given mode, but also specific vectors in a specific mode. Each submap specifies the proper floor and residue instance number to use for decoding that submap's spectral floor and spectral residue vectors.
As an example:
Assume a Vorbis stream that contains six channels in the standard 5.1 format. The sixth channel, as is normal in 5.1, is bass only. Therefore it would be wasteful to encode a full-spectrum version of it as with the other channels. The submapping mechanism can be used to apply a full range floor and

residue encoding to channels 0 through 4, and a bass-only representation to the bass channel, thus saving space. In this example, channels 0-4 belong to submap 0 (which indicates use of a full-range floor) and channel 5 belongs to submap 1, which uses a bass-only representation.

5.4 Floor

Vorbis encodes a spectral ’floor’ vector for each PCM channel. This vector is a low-resolution representation of the audio spectrum for the given channel in the current frame, generally used akin to a whitening filter. It is named a ’floor’ because the Xiph.Org reference encoder has historically used it as a unit-baseline for spectral resolution.
A floor encoding may be of two types. Floor 0 uses a packed LSP representation on a dB amplitude scale and Bark frequency scale. Floor 1 represents the curve as a piecewise linear interpolated representation on a dB amplitude scale and linear frequency scale. The two floors are semantically interchangeable in encoding/decoding. However, floor type 1 provides more stable inter-frame behavior, and so is the preferred choice in all coupled-stereo and high bitrate modes. Floor 1 is also considerably less expensive to decode than floor 0.
Floor 0 is not to be considered deprecated, but it is of limited modern use. No known Vorbis encoder past Xiph.org’s own beta 4 makes use of floor 0.
The values coded/decoded by a floor are both compactly formatted and make use of entropy coding to save space. For this reason, a floor configuration generally refers to multiple codebooks in the codebook component list. Entropy coding is thus provided as an abstraction, and each floor instance may choose from any and all available codebooks when coding/decoding.

5.5 Residue

The spectral residue is the fine structure of the audio spectrum once the floor curve has been subtracted out. In simplest terms, it is coded in the bitstream using cascaded (multi-pass) vector quantization according to one of three specific packing/coding algorithms numbered 0 through 2. The packing algorithm details are configured by residue instance. As with the floor components, the final VQ/entropy encoding is provided by external codebook instances and each residue instance may choose from any and all available codebooks.

5.6 Codebooks

Codebooks are a self-contained abstraction that perform entropy decoding and, optionally, use the entropy-decoded integer value as an offset into an index of output value vectors, returning the indicated vector of values.
The entropy coding in a Vorbis I codebook is provided by a standard Huffman binary tree representation. This tree is tightly packed using one of several methods, depending on whether codeword lengths are ordered or unordered, or the tree is sparse.
The codebook vector index is similarly packed according to index characteristic. Most commonly, the vector index is encoded as a single list of values of possible values that are then permuted into a list of n-dimensional rows (lattice VQ).

6 High-level Decode Process

6.1 Decode setup

Before decoding can begin, a decoder must initialize using the bitstream headers matching the stream to be decoded. Vorbis uses three header packets; all are required, in-order, by this specification. Once set up, decode may begin at any audio packet belonging to the Vorbis stream. In Vorbis I, all packets after the three initial headers are audio packets.
The header packets are, in order, the identification header, the comments header, and the setup header.

6.1.1 Identification Header

The identification header identifies the bitstream as Vorbis, Vorbis version, and the simple audio characteristics of the stream such as sample rate and number of channels.

6.1.2 Comment Header

The comment header includes user text comments ["tags"] and a vendor string for the application/library that produced the bitstream. The encoding of the comment header is described within this document; the proper use of the comment fields is described in the Ogg Vorbis comment field specification.

6.1.3 Setup Header

The setup header includes extensive CODEC setup information as well as the complete VQ and Huffman codebooks needed for decode.

6.2 Decode Procedure

The decoding and synthesis procedure for all audio packets is fundamentally the same.

1. decode packet type flag
2. decode mode number
3. decode window shape [long windows only]
4. decode floor
5. decode residue into residue vectors
6. inverse channel coupling of residue vectors
7. generate floor curve from decoded floor data
8. compute dot product of floor and residue, producing audio spectrum vector
9. inverse monolithic transform of audio spectrum vector, always an MDCT in Vorbis I
10. overlap/add left-hand output of transform with right-hand output of previous frame
11. store right hand-data from transform of current frame for future lapping.
12. if not first frame, return results of overlap/add as audio result of current frame

Note that clever rearrangement of the synthesis arithmetic is possible; as an example, one can take advantage of symmetries in the MDCT to store the right-hand transform data of a partial MDCT for a 50% inter-frame buffer space savings, and then complete the transform later before overlap/add with the next frame. This optimization produces entirely equivalent output and is naturally perfectly legal. The decoder must be *entirely mathematically equivalent* to the specification, it need not be a literal semantic implementation.

6.2.1 Packet type decode

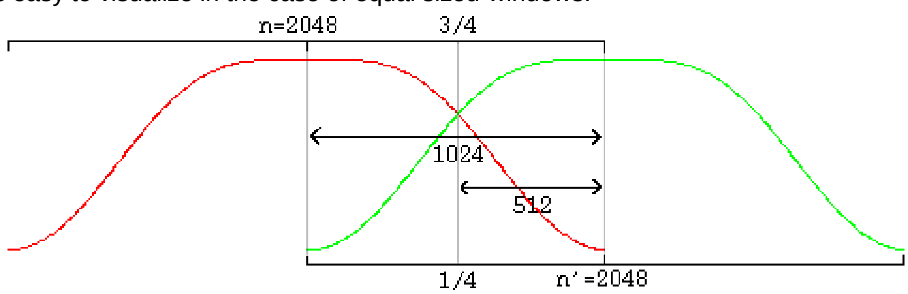
Vorbis I uses four packet types. The first three packet types mark each of the three Vorbis headers described above. The fourth packet type marks an audio packet. All others packet types are reserved; packets marked with a reserved flag type should be ignored.
Following the three header packets, all packets in a Vorbis I stream are audio. The first step of audio packet decode is to read and verify the packet type; *a non-audio packet when audio is expected indicates stream corruption or a non-compliant stream. The decoder must ignore the packet and not attempt decoding it to audio*.

6.2.2 Mode decode

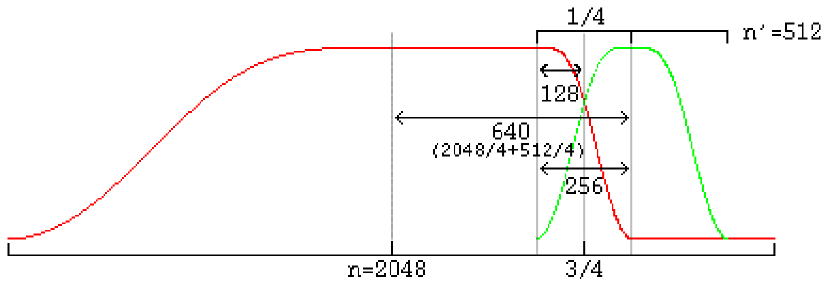
Vorbis allows an encoder to set up multiple, numbered packet 'modes', as described earlier, all of which may be used in a given Vorbis stream. The mode is encoded as an integer used as a direct offset into the mode instance index.

6.2.3 Window shape decode [long windows only]

Vorbis frames may be one of two PCM sample sizes specified during codec setup. In Vorbis I, legal frame sizes are powers of two from 64 to 8192 samples. Aside from coupling, Vorbis handles channels as independent vectors and these frame sizes are in samples per channel.
Vorbis uses an overlapping transform, namely the MDCT, to blend one frame into the next, avoiding most inter-frame block boundary artifacts. The MDCT output of one frame is windowed according to MDCT requirements, overlapped 50% with the output of the previous frame and added. The window shape assures seamless reconstruction.
This is easy to visualize in the case of equal sized-windows:



And slightly more complex in the case of overlapping unequal sized windows:



In the unequal-sized window case, the window shape of the long window must be modified for seamless lapping as above. It is possible to correctly infer window shape to be applied to the current window from knowing the sizes of the current, previous and next window. It is legal for a decoder to use this method; However, in the case of a long window (short windows require no modification), Vorbis also codes two flag bits to specify pre- and post- window shape. Although not strictly necessary for function, this minor redundancy allows a packet to be fully decoded to the point of lapping entirely independently of any other packet, allowing easier abstraction of decode layers as well as allowing a greater level of easy parallelism in encode and decode.
A description of valid window functions for use with an inverse MDCT can be found in the paper _The use of multirate filter banks for coding of high quality digital audio_, by T. Sporer, K. Brandenburg and B. Edler. Vorbis windows all use the slope function y=sin(2PI*sin^2(x/n)).

6.2.4 floor decode

Each floor is encoded/decoded in channel order, however each floor belongs to a 'submap' that specifies which floor configuration to use. All floors are decoded before residue decode begins.

6.2.5 residue decode

Although the number of residue vectors equals the number of channels, channel coupling may mean that the raw residue vectors extracted during decode do not map directly to specific channels. When channel coupling is in use, some vectors will correspond to coupled magnitude or angle. The coupling relationships are described in the codec setup and may differ from frame to frame, due to different mode numbers.
Vorbis codes residue vectors in groups by submap; the coding is done in submap order from submap 0 through n-1. This differs from floors which are coded using a configuration provided by submap number, but are coded individually in channel order.

6.2.6 inverse channel coupling

A detailed discussion of stereo in the Vorbis codec can be found in the document _Stereo Channel Coupling in the Vorbis CODEC_. Vorbis is not limited to only stereo coupling, but the stereo document also gives a good overview of the generic coupling mechanism.
Vorbis coupling applies to pairs of residue vectors at a time; decoupling is done in-place a pair at a time in the order and using the vectors specified in the current mapping configuration. The decoupling operation is the same for all pairs, converting square polar representation (where one vector is magnitude and the second angle) back to Cartesian representation.
After decoupling, in order, each pair of vectors on the coupling list in, the resulting residue vector represents the fine spectral detail of each output channel.

6.2.7 generate floor curve

The decoder may choose to generate the floor curve at any appropriate time. It is reasonable to generate the output curve when the floor data is decoded from the raw packet, or it can be generated after inverse coupling and applied to the spectral residue directly, combining generation and the dot product into one step and eliminating some working space.
Both floor 0 and floor 1 generate a linear-range, linear-domain output vector to be multiplied (dot product) by the linear-range, linear-domain spectral residue.

6.2.8 compute floor/residue dot product

This step is straightforward; for each output channel, the decoder multiplies the floor curve and residue vectors element by element, producing the finished audio spectrum of each channel.
One point is worth mentioning about this dot product; a common mistake in a fixed point implementation might be to assume that a 32 bit fixed-point representation for floor and residue and direct multiplication of the vectors is sufficient for acceptable spectral depth in all cases because it happens to mostly work with the current Xiph.Org reference encoder.
However, floor vector values can span ~140dB (~24 bits unsigned), and the audio spectrum vector should represent a minimum of 120dB (~21 bits with sign), even when output is to a 16 bit PCM device. For the residue vector to represent full scale if the floor is nailed to -140dB, it must be able to span 0 to +140dB. For the residue vector to reach full scale if the floor is nailed at 0dB, it must be able to represent -140dB to +0dB. Thus, in order to handle full range dynamics, a residue vector may span -140dB to +140dB entirely within spec. A 280dB range is approximately 48 bits with sign; thus the residue vector must be able to represent a 48 bit range and the dot product must be able to handle an effective 48 bit times 24 bit multiplication. This range may be achieved using large (64 bit or larger) integers, or implementing a movable binary point representation.

6.2.9 inverse monolithic transform (MDCT)

The audio spectrum is converted back into time domain PCM audio via an inverse Modified Discrete Cosine Transform (MDCT). A detailed description of the MDCT is available in the paper _The use of multirate filter banks for coding of high quality digital audio_, by T. Sporer, K. Brandenburg and B. Edler.
Note that the PCM produced directly from the MDCT is not yet finished audio; it must be lapped with surrounding frames using an appropriate window (such as the Vorbis window) before the MDCT can be considered orthogonal.

6.2.10 overlap/add data

Windowed MDCT output is overlapped and added with the right hand data of the previous window such that the 3/4 point of the previous window is aligned with the 1/4 point of the current window (as illustrated in the window overlap diagram). At this point, the audio data between the center of the previous frame and the center of the current frame is now finished and ready to be returned.

6.2.11 cache right hand data

The decoder must cache the right hand portion of the current frame to be lapped with the left hand portion of the next frame.

6.2.12 return finished audio data

The overlapped portion produced from overlapping the previous and current frame data is finished data to be returned by the decoder. This data spans from the center of the previous window to the center of the current window. In the case of same-sized windows, the amount of data to return is one-half block consisting of and only of the overlapped portions. When overlapping a short and long window, much of the returned range is not actually overlap. This does not damage transform orthogonality. Pay attention however to returning the correct data range; the amount of data to be returned is:
window_blocksize(previous_window)/4+window_blocksize(current_window)/4 from the center of the previous window to the center of the current window.
Data is not returned from the first frame; it must be used to ’prime’ the decode engine. The encoder accounts for this priming when calculating PCM offsets; after the first frame, the proper PCM output offset is ’0’ (as no data has been returned yet).