The Speex Codec Manual
(draft for Speex 1.0beta4)

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Disclaimer

This document is meant to provide useful information on the Speex codec but there is absolutely no warranty regarding usefulness or correctness of the information provided. Also, some techniques used in Speex are said to be "similar" to techniques used in known codecs. This should not be understood as an acknowledgment that Speex is using any patented algorithm used in these codecs, but merely that comprehension of Speex can be facilitated by thinking that the principles of operation are the same or similar. Of course, there's also the obligatory "all trademarks are property of their respective owner".

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1 Introduction to Speex

The Speex project (http://www.speex.org/) has been started because there was a need for a speech codec that was open-source and free from software patents. These are essential conditions for being used by any open-source software. There is already Vorbis that does general audio, but it is not really suitable for speech. Also, unlike many other speech codecs, Speex is not targeted at cell phones (not many open-source cell phones anyway :-)) but rather voice over IP (VoIP) and file-based compression.

As design goals, we wanted to have a codec that would allowed both very good quality speech and low bit-rate (unfortunately not at the same time!), which led us to developing a codec with multiple bit-rates. Of course very good quality also meant we had to do wideband (16 kHz sampling rate) in addition to narrowband (telephone quality, 8 kHz sampling rate).

Designing for VoIP instead of cell phone use means that Speex must be robust to lost packets, but not to corrupted ones since packets either arrive unaltered or don't arrive at all. Also, the idea was to have a reasonnable complexity and memory requirement without compromising too much on the efficiency of the codec.

All this led us to the choice of CELP as the encoding technique to use for Speex. One of the main reasons is that CELP has long proved that it could do the job and scale well to both low bit-rates (think DoD CELP @ 4.8 kbps) and high bit-rates (think G.728 @ 16 kbps).

The main characteristics can be summerized as follows:

- Free software/open-source, patent and royalty-free
- Integration of wideband and narrowband in the same bit-stream
- Wide range of bit-rates available
- Dynamic bit-rate switching and Variable Bit-Rate (VBR)
- Voice Activity Detection (VAD, integrated with VBR)
- Variable complexity
- Ultra-wideband mode at 32 kHz (up to 48 kHz)
- Intensity stereo encoding option

2 Introduction to CELP Coding

Speex is based on CELP, which stands for Code Excited Linear Prediction. This section attempts to introduce the principles behind CELP, so if you are already familiar with CELP, you can safely skip to section 3. The CELP technique is based on three ideas:

1. The use of a linear prediction (LP) model to model the vocal tract
2. The use of (adaptive and fixed) codebook entries as input (excitation) of the LP model
3. The search performed in closed-loop in a "perceptually weighted domain"

2.1 Linear Prediction (LPC)

Linear prediction is at the base of may speech coding techniques, including CELP. The idea behind it is to predict the signal $x(n)$ using a linear combination of its past samples:

$$y[n]=\sum_{i=1}^{N}a_i x[n-i]$$

where $y[n]$ is the linear prediction of $x[n]$. The prediction error is thus given by:

$$e[n]=x[n]-y[n]=x[n]-\sum_{i=1}^{N}a_i x[n-i]$$

The goal of the LPC analysis is to find the best prediction coefficients a_i which minimize the quadratic error function:

$$E=\sum_{n=0}^{L-1}[e[n]]^2=\sum_{n=0}^{L-1}\left[x[n]-\sum_{i=1}^{N}a_i x[n-i]\right]^2$$

That can be done by making all derivatives $\frac{\partial E}{\partial a_i}$ equal to zero:

$$\frac{\partial E}{\partial a_i}=\frac{\partial}{\partial a_i}\sum_{n=0}^{L-1}\left[x[n]-\sum_{i=1}^{N}a_i x[n-i]\right]^2=0$$

The a_i filter coefficients are computed using the Levinson-Durbin algorithm, which starts from the auto-correlation $R(m)$ of the signal $x[n]$.

$$R(m)=\sum_{i=0}^{N-1}x[i]x[i-m]$$

For an order N filter, we have:

$$\mathbf{R} = \begin{bmatrix} R(0) & R(1) & \cdots & R(N-1) \\ R(1) & R(0) & \cdots & R(N-2) \\ \vdots & \vdots & \ddots & \vdots \\ R(N-1) & R(N-2) & \cdots & R(0) \end{bmatrix}$$

$$\mathbf{r} = \begin{bmatrix} R(1) \\ R(2) \\ \vdots \\ R(N) \end{bmatrix}$$

The filter coefficients a_i are found by solving the system $\mathbf{Ra} = \mathbf{r}$. What the Levinson-Durbin algorithm does here is making the solution to the problem $\mathcal{O}\left(N^2\right)$ instead of $\mathcal{O}\left(N^3\right)$ by exploiting the fact that matrix $\mathbf{R}$ is toeplitz hermitian. Also, it can be proved that all the roots of $A(z)$ are withing the unit circle, which means that $1/A(z)$ is always stable. This is in theory; in practice because of finite precision, there are two commonly used techniques to make sure we have a stable filter. First, we multiply $R(0)$ by a number slightly above one (such as 1.0001), which is equivalent to adding noise to the signal. Also, we can apply a window the the auto-correlation, which is equivalent to filtering in the frequency domain, reducing sharp resonances.

The linear prediction model represents each speech sample as linear combination of past samples, plus an error signal called the excitation (or residual).

$$x[n] = \sum_{i=1}^{N} a_i x[n-i] + e[n]$$

In the z-domain, this can be expressed as

$$x(z) = \frac{1}{A(z)}\, e(z)$$

where $A(z)$ is defined as

$$A(z) = 1 - \sum_{i=1}^{N} a_i z^{-i}$$

We usually refer to $A(z)$ as the analysis filter and $1/A(z)$ as the synthesis filter. The whole process is called short-term prediction as it predicts the signal $x[n]$ using a prediction using only the N past samples, where N is usually around 10.

Because LPC coefficients have very little robustness to quantization, they are converted to Line Spectral Pair (LSP) coefficients which have a much better behaviour with quantization, one of them being that it's easy to keep the filter stable.

2.2 Pitch Prediction

During voiced segments, the speech signal is periodic, so it is possible to take advantage of that property by approximating the excitation signal $e[n]$ by a gain times the past of the excitation:

$$e[n] \simeq p[n] = \beta e[n-T]$$

where T is the pitch period, β is the pitch gain and $c(n)$ is taken from the *innovation codebook*. We call that long-term prediction since the excitation is predicted from $e[n-T]$ with $T \gg N$.

2.3 Innovation Codebook

The final excitation $e[n]$ will be the sum of the pitch prediction and an *innovation* signal $c[n]$ taken from a fixed codebook.

$$e[n] = p[n] + c[n] = \beta e[n-T] + c[n]$$

This is where most of the bits in a CELP codec are allocated. It represents the information that couldn't be obtained either from linear prediction or pitch prediction. In the z-domain we can represent the final signal $X(z)$ as

$$X(z) = \frac{C(z)}{A(z)\left(1-\beta z^{-T}\right)}$$

2.4 Analysis-by-Synthesis and Error Weighting

Most (if not all) modern audio codecs attempt to "shape" the noise so that it appears mostly in the frequency regions where the ear cannot detect it. For example, the ear is more tolerant to noise in parts of the spectrum that are louder and *vice versa*. That's why instead of minimizing the simple quadratic error

$$E = \sum_n \left(x[n] - \overline{x}[n]\right)^2$$

where $\overline{x}[n]$ is the encoder signal, we minimize the error for the perceptually weighted signal

$$X_w(z) = W(z)X(z)$$

where $W(z)$ is the weighting filter, usually of the form

$$W(z) = \frac{A\left(\frac{z}{\gamma_1}\right)}{A\left(\frac{z}{\gamma_2}\right)} \tag{1}$$

with control parameters $\gamma_1 > \gamma_2$. If the noise is white in the perceptually weighted domain, then in the signal domain its spectral shape will be of the form

$$A_{noise}(z) = \frac{1}{W(z)} = \frac{A\left(\frac{z}{\gamma_2}\right)}{A\left(\frac{z}{\gamma_1}\right)}$$

If a filter $A(z)$ has (complex) poles at p_i in the z-plane, the filter $A(z/\gamma)$ filter will have its poles at $p_i^{'} = \gamma p_i$, making it a flatter version of $A(z)$.

3 Speex narrowband mode

This section looks at how Speex works for narrowband (8 kHz sampling rate) operation. The frame size for this mode is 20 ms, corresponding to 160 samples. Each frame is also subdivided into 4 sub-frames of 40 samples each.

Also many design decisions were based on the original goals and assumptions:

- Minimizing the amount of information extracted from past frames (for robustness to packet loss)
- Dynamically-selectable codebooks (LSP, pitch and innovation)
- G.728-like fixed codebooks (without backward-adaptive gains because of patent issues)

3.1 LPC Analysis

An LPC analysis is first performed on a (asymetric Hamming) window that spans all the current frame and half a frame in advance. The LPC coefficients are then converted to Line Spectral Pair (LSP), a representation that is more robust to quantization. The LSP's are considered to be associated to the 4^{th} sub-frames and the LSP's associated to the first 3 sub-frames are linearly interpolated using the current and previous LSP's.

The LSP's are encoded using 30 bits for higher quality modes and 18 bits for lower quality, through the use of a multi-stage split-vector quantizer. For the lower quality modes, the 10 coefficients are first quantized with 6 bits and the error is then divided in two 5-coefficient sub-vectors. Each of them is quantized with 6 bits, for a total of 18 bits. For the higher quality modes, the remaining error on both sub-vectors is further quantized with 6 bits each, for a total of 30 bits.

The perceptual weighting filter $W(z)$ used by Speex is derived from the LPC filter $A(z)$ and corresponds to the one described by eq. 1 with $\gamma_1 = 0.9$ and $\gamma_2 = 0.6$. We can use the unquantized $A(z)$ filter since the weighting filter is only used in the encoder.

3.2 Pitch Prediction (adaptive codebook)

Speex uses a 3-tap prediction for pitch. That is, the pitch prediction signal $p[n]$ is obtained by the past of the excitation by:

$$p[n] = \beta_0 e[n - T - 1] + \beta_1 e[n - T] + \beta_2 e[n - T + 1]$$

where T is the pitch period and the β_i are the prediction (filter) taps. It is worth noting that when the pitch is smaller than the sub-frame size, we repeat the excitation at a period T. For example, when $n - T + 1$, we use $n - 2T + 1$ instead. The period and quantized gains are determined in closed loop. In most modes, the pitch period is encoded with 7 bits in the $[17, 144]$ range and the β_i coefficients are vector-quantized using 7 bits (15 kbps narrowband and above) at higher bit-rates and 5 bits at lower bit-rates (11 kbps narrowband and below).

3.3 Innovation Codebook

In Speex, the innovation signal is quantized using shape-only vector quantization (VQ). That means that the codebooks that are used represent both the shape and the gain at the same time. This save many bits that would otherwise be allocated for a separate gain at the price of a slight increase in complexity. Except for the absence of (backward-adaptive) gain, the technique used in Speex is similar to G.728 (LD-CELP). However since we do not have a low-delay constraint, the search can be made more "global" and make use of the whole information available for a sub-frame.

3.4 Bit allocation

There are 7 different narrowband bit-rates defined for Speex, ranging from 200 bps to 18.15 kbps, although the modes below 5.9 kbps should not be used for speech. The bit-allocation for each mode is detailed in table 1. Each frame starts with the mode ID encoded with 4 bits which allows a range from 0 to 15, though only the first 7 values are used (the others are reserved). The parameters are listed in the table in the order they are packed in the bit-stream. All frame-based parameters are packed before sub-frame parameters. The parameters for a certain sub-frame are all packed before the following sub-frame is packed. Note that the "OL" in the parameter description means the the parameter is an open loop estimation based on the whole frame.

Parameter	Update rate	0	1	2	3	4	5	6	7
Wideband bit	frame	1	1	1	1	1	1	1	1
Mode ID	frame	4	4	4	4	4	4	4	4
LSP	frame	0	18	18	18	18	30	30	30
OL pitch	frame	0	7	7	0	0	0	0	0
OL pitch gain	frame	0	4	0	0	0	0	0	0
OL Exc gain	frame	0	5	5	5	5	5	5	5
Fine pitch	sub-frame	0	0	0	7	7	7	7	7
Pitch gain	sub-frame	0	0	5	5	5	7	7	7
Innovation gain	sub-frame	0	1	0	1	1	3	3	3
Innovation VQ	sub-frame	0	0	16	20	35	48	64	96
Total	frame	5	43	119	160	220	300	364	492

Table 1: Bit allocation for narrowband modes

So far, no MOS (Mean Opinion Score) subjective evaluation has been performed for Speex. In order to give an idea of the quality achivable with it, table 2 presents my own subjective opinion on it. It sould be noted that different people will perceive the quality differently and that the person that designed the codec often has a bias (one way or another) when it comes to subjective evaluation. Last thing, it should be noted that for most codecs (including Speex) encoding quality sometimes varies depending on the input. Note that the complexity is only approximate (withing 0.5 mflops and using the lowers complexity setting). Decoding requires approximately 0.5 mflops in most modes (1 mflops with perceptual enhancement).

Mode	Bit-rate (bps)	mflops	Quality/description
0	250	N/A	No sound (VBR only)
1	2,150	6	Vocoder (mostly for comfort noise)
2	5,950	9	Very noticeable artifacts/noise, good intelligibility
3	8,000	10	Artifacts/noise sometimes noticeable
4	11,000	14	Artifacts usually noticeable only with headphones
5	15,000	11	Need good headphones to tell the difference
6	18,200	17.5	Hard to tell the difference even with good headphones
7	24,600	14.5	Completely transparent for voice, good quality music
8	N/A	N/A	reserved
9	N/A	N/A	reserved
10	N/A	N/A	reserved
11	N/A	N/A	reserved
12	N/A	N/A	reserved
13	N/A	N/A	Application-defined, interpreted by callback or skipped
14	N/A	N/A	Speex in-band signaling
15	N/A	N/A	Terminator code

Table 2: Quality versus bit-rate

3.5 Perceptual enhancement

This part of the codec only applies to the decoder and can even be changed without affecting inter-operability. For that reason, the implementation provided and described here should only be considered as a reference implementation. The enhancement system is devided in two parts. First, the synthesis filter $S(z) = 1/A(z)$ is replaced by an enhanced filter

$$S'(z) = \frac{A\left(z/a_2\right)A\left(z/a_3\right)}{A\left(z\right)A\left(z/a_1\right)}$$

where a_1 and a_2 depend on the mode in use and $a_3 = \frac{1}{r}\left(1 - \frac{1-ra_1}{1-ra_2}\right)$ with $r = .9$. The second part of the enhancement consists of using a comb filter to enhance the pitch in the excitation domain.

4 Speex wideband mode (sub-band CELP)

For wideband, the Speex approach uses a *q*uadrature *m*irror *f*ilter (QMF) to split the band in two. The 16 kHz signal is thus divided into two 8 kHz signals, one representing the low band (0-4 kHz), the other the high band (4-8 kHz). The low band is encoded with the narrowband mode described in section 3 in such a way that the resulting "embedded narrowband bit-stream" can also be decoded with the narrowband decoder. Since the low band encoding has already been described only the high band encoding is described in this section.

4.1 Linear Prediction

The linear prediction part used for the high-band is very similar to what is done for narrowband. The only difference is that we use only 12 bits to encode the high-band LSP's using a multi-stage vector quantizer (MSVQ). The first level quantizes the 10 coefficients with 6 bits and the error is then quantized using 6 bits too.

4.2 Pitch Prediction

That part is easy: there's no pitch prediction for the high-band. There are two reasons for that. First, there is usually little harmonic structure in this band (above 4 kHz). Second, it would be very hard to implement since the QMF folds the 4-8 kHz band into 4-0 kHz (reversing the frequency axis), which means that the location of the harmonics are no longer at multiples of the fundamental (pitch).

4.3 Excitation Quantization

The high-band excitation is coded in the same way as for narrowband.

4.4 Bit allocation

For the wideband mode, all the narrowband frame is packed before the high-band is encoded. The narrowband part of the bit-stream is as defined in table 1. The high-band follows, as described in table 3. This also means that a wideband frame may be correctly decoded by a narrowband decoder with the only caveat that if more than one frame is packed in the same packet, the decoder will need to skip the high-band parts in order to sync with the bit-stream.

Parameter	Update rate	0	1	2	3	4
Wideband bit	frame	1	1	1	1	1
Mode ID	frame	3	3	3	3	3
LSP	frame	0	12	12	12	12
Excitation gain	sub-frame	0	5	4	4	4
Excitation VQ	sub-frame	0	0	20	40	80
Total	frame	4	36	112	192	352

Table 3: Bit allocation for high-band in wideband mode

5 Command-line encoder/decoder

The base Speex distribution includes a command-line encoder (*speexenc*) and decoder (*speexdec*).

5.1 *speexenc*

The encoder takes the following options:

–narrowband (-n) Tell Speex to treat the input as narrowband (8 kHz). This is the default

–wideband (-w) Tell Speex to treat the input as wideband (16 kHz)

–ultra-wideband (-u) Tell Speex to treat the input as "ultra-wideband" (32 kHz)

–quality n Set the encoding quality (0-10), default is 3

–bitrate n Encoding bit-rate (use bit-rate n or lower)

–vbr Enable VBR (Variable Bit-Rate), disabled by default

–abr n Enable ABR (Average Bit-Rate) at n kbps, disabled by default

–vad Enable VAD (Voice Activity Detection), disabled by default

–dtx Enable DTX (Discontinuous Transmission), disabled by default

–nframes n Pack n frames in each Ogg packet (this saves space at low bit-rates)

–comp n Set encoding speed/quality tradeoff. The higher the value of n, the slower the encoding (default is 3)

-V Verbose operation, print bit-rate currently in use

–help (-h) Print the help

–version (-v) Print version information

Speex comments

–comment Add the given string as an extra comment. This may be used multiple times.

–author Author of this track.

–title Title for this track.

Raw input options

–rate n Sampling rate for raw input

–stereo Consider raw input as stereo

–le Raw input is little-endian

–be Raw input is big-endian

–8bit Raw input is 8-bit unsigned

–16bit Raw input is 16-bit signed

5.2 *speexdec*

The decoder takes the following options:

–enh enable post-filter (default)

–no-enh disable post-filter

–force-nb Force decoding in narrowband

–force-wb Force decoding in wideband

–force-uwb Force decoding in ultra-wideband

–mono Force decoding in mono

–stereo Force decoding in stereo

–rate n For decoding at n Hz sampling rate

–packet-loss n Simulate n % rando m packet loss

-V Verbose operation, print bit-rate currently in use

–help (-h) Print the help

–version (-v) Print version information

6 Programming with Speex (the libspeex API)

6.1 Encoding

In order to encode speech using Speex, you first need to:

```
#include <speex.h>
```

You then need to declare a Speex bit-packing struct

```
SpeexBits bits;
```

and a Speex encoder state

```
void *enc_state;
```

The two are initialized by:

```
speex_bits_init(&bits);
enc_state = speex_encoder_init(&speex_nb_mode);
```

For wideband coding, *speex_nb_mode* will be replaced by *speex_wb_mode*. In most cases, you will need to know the frame size used by the mode you are using. You can get that value in the *frame_size* variable with:

```
speex_encoder_ctl(enc_state,SPEEX_GET_FRAME_SIZE,&frame_size);
```

Once the initialization is done, for every input frame:

```
speex_bits_reset(&bits);
speex_encode(enc_state, input_frame, &bits);
nbBytes = speex_bits_write(&bits, byte_ptr, MAX_NB_BYTES);
```

where *input_frame* is a *(float *)* pointing to the beginning of a speech frame, *byte_ptr* is a *(char *)* where the encoded frame will be written, *MAX_NB_BYTES* is the maximum number of bytes that can be written to *byte_ptr* without causing an overflow and *nbBytes* is the number of bytes actually written to *byte_ptr* (the encoded size in bytes). Before calling speex_bits_write, it is possible to find the number of bytes that need to be written by calling `speex_bits_nbytes(&bits)`, which returns a number of bytes.

After you're done with the encoding, free all resources with:

```
speex_bits_destroy(&bits);
speex_encoder_destroy(enc_state);
```

That's about it for the encoder.

6.2 Decoding

In order to encode speech using Speex, you first need to:

```
#include <speex.h>
```

You then need to declare a Speex bit-packing struct

```
SpeexBits bits;
```

and a Speex encoder state

```
void *dec_state;
```

The two are initialized by:

```
speex_bits_init(&bits);
dec_state = speex_decoder_init(&speex_nb_mode);
```

For wideband decoding, *speex_nb_mode* will be replaced by *speex_wb_mode*. You can get that value in the *frame_size* variable with:

```
speex_decoder_ctl(dec_state, SPEEX_GET_FRAME_SIZE, &frame_size);
```

There is also a parameter that can be set for the decoder: whether or not to use a perceptual post-filter. This can be set by:

```
speex_decoder_ctl(dec_state, SPEEX_SET_PF, &pf);
```

where pf is an int that with value 0 to have the post-filter disabled and 1 to have it enabled.

Again, once the decoder initialization is done, for every input frame:

```
speex_bits_read_from(&bits, input_bytes, nbBytes);
speex_decode(st, &bits, output_frame);
```

where input_bytes is a *(char *)* containing the bit-stream data received for a frame, *nbBytes* is the size (in bytes) of that bit-stream, and *output_frame* is a *(float *)* and points to the area where the decoded speech frame will be written. A NULL value as the first argument indicates that we don't have the bits for the current frame. When a frame is lost, the Speex decoder will do its best to "guess" the correct signal.

After you're done with the decoding, free all resources with:

```
speex_bits_destroy(&bits);
speex_decoder_destroy(dec_state);
```

6.3 Codec Options (speex_*_ctl)

The Speex encoder and decoder support many options and requests that can be accessed through the *speex_encoder_ctl* and *speex_decoder_ctl* functions. These functions are similar the the *ioctl* system call and their prototypes are:

```
void speex_encoder_ctl(void *encoder, int request, void *ptr);
void speex_decoder_ctl(void *encoder, int request, void *ptr);
```

The different values of request allowed are (note that some only apply to the encoder or the decoder):

SPEEX_SET_ENH** Set perceptual enhancer to on (1) or off (0) (integer)

SPEEX_GET_ENH** Get perceptual enhancer status (integer)

SPEEX_GET_FRAME_SIZE Get the frame size used for the current mode (integer)

SPEEX_SET_QUALITY* Set the encoder speech quality (integer 0 to 10)

SPEEX_GET_QUALITY* Get the current encoder speech quality (integer 0 to 10)

SPEEX_SET_MODE*†

SPEEX_GET_MODE*†

SPEEX_SET_LOW_MODE*†

SPEEX_GET_LOW_MODE*†

SPEEX_SET_HIGH_MODE*†

SPEEX_GET_HIGH_MODE*†

SPEEX_SET_VBR* Set variable bit-rate (VBR) to on (1) or off (0) (integer)

SPEEX_GET_VBR* Get variable bit-rate (VBR) status (integer)

SPEEX_SET_VBR_QUALITY* Set the encoder VBR speech quality (integer 0 to 10)

SPEEX_GET_VBR_QUALITY* Get the current encoder VBR speech quality (integer 0 to 10)

SPEEX_SET_COMPLEXITY* Set the CPU resources allowed for the encoder

SPEEX_GET_COMPLEXITY* Get the CPU resources allowed for the encoder

SPEEX_SET_BITRATE* Set the bit-rate to use to the closest value not exceeding the parameter (integer in bps)

SPEEX_GET_BITRATE Get the current bit-rate in use (integer in bps)

SPEEX_SET_SAMPLING_RATE Set real sampling rate (is used to determine real bit-rate)

SPEEX_GET_SAMPLING_RATE Get real sampling rate (is used to determine real bit-rate)

***** applies only to the encoder

****** applies only to the decoder

† normally only used internally

6.4 Mode queries

Speex modes have a querry system similar to the speex_encoder_ctl and speex_decoder_ctl calls. Since modes are read-only, it is only possible to get information about a particular mode. The function used to do that is:

```
void speex_mode_query(SpeexMode *mode, int request, void *ptr);
```

The admissible values for request are (unless otherwise note, the values are returned through *ptr*):

SPEEX_MODE_FRAME_SIZE Get the frame size (in samples) for the mode

SPEEX_SUBMODE_BITRATE Get the bit-rate for a submode number specified throught *ptr*.

6.5 Packing and in-band signalling

Sometimes it is desirable to pack more than one frame per packet (or other basic unit of storage). The proper way to do it is to call speex_encode N times before writing the stream with speex_bits_write. In cases where the number of frames is not determined by an out-of-band mechanism, it is possible to include a terminator code. That terminator consists of the code 15 (decimal) encoded with 5 bits, as shown in figure 2.

It is also possible to send in-band "messages" to the other side. All these messages are encoded as a "pseudo-frame" of mode 14 which contain a 4-bit message type code, followed by the message. Table 4 lists the available codes, their meaning and the size of the message that follow. Most of these messages are requests that are sent to the encoder or decoder on the other end, which is free to comply or ignore them. By default, all in-band messages are ignored.

code	Size (bits)	Content
0	1	Asks decoder to set perceptual enhancement off (0) or on(1)
1	1	reserved
2	4	Asks encoder to switch to mode N
3	4	Asks encoder to switch to mode N for low-band
4	4	Asks encoder to switch to mode N for high-band
5	4	Asks encoder to switch to quality N for VBR
6	4	Request acknowloedge (0=no, 1=all, 2=only for in-band data)
7	4	Asks encoder to set VBR off (0), on(1), VAD(2), DTX(3)
8	8	Transmit (8-bit) character to the other end
9	8	Intensity stereo information
10	16	Announce maximum bit-rate acceptable (N in bytes/second)
11	16	reserved
12	32	Acknowledge receiving packet N
13	32	reserved
14	64	reserved
15	64	reserved

Table 4: In-band signalling codes

Finally, applications may define custom in-band messages using mode 13. The size of the message in bytes is encoded with 5 bits, so that the decoder can skip it if it doesn't know how to interpret it.

7 Formats and standards

Speex can encode speech in both narrowband and wideband and provides different bit-rates. However not all features must be supported by a certain implementation or device. In order to be said "Speex compatible" (whatever that means), an implementation must implement at least a basic set of features.

At the minimum, all narrowband modes of operation MUST be supported at the decoder. This includes the decoding of a wideband bit-stream by the narrowband decoder[1]. If present, a wideband decoder MUST be able to decode a narrowband stream, and MAY either be able to decode all wideband modes or be able to decode the embedded narrowband part of all modes (which includes ignoring the high-band bits).

For encoders, at least one narrowband or wideband mode MUST be supported. The main reason why all encoding modes do not have to be supported is that some platforms may not be able to handle the complexity of encoding in some modes.

7.1 RTP Payload Format

The latest RTP payload draft can be found at `http://www.speex.org/drafts/latest`. We are (2002/11/11) about to send the latest draft to the IETF for comments.

7.2 MIME Type

Speex will use the MIME type `audio/speex`. We will apply for that type in the near future.

7.3 Ogg file format

Speex bit-streams can be stored in Ogg files. In this case, the first packet of the Ogg file contains the Speex header described in table 5. All integer fields in the headers are stored as little-endian. The `speex_string` field must contain the "`Speex   `" (with 3 training spaces), which identifies the bit-stream. The next field, `speex_version` contains the version of Speex that encoded the file. For now, refer to speex_header.[ch] for more info. The *beginning of stream* (`b_o_s`) flag is set to 1 for the header. The header packet has `packetno=0` and `granulepos=0`.

The second packet contains the Speex comment header. The format used is the Vorbis comment format described here: http://www.xiph.org/ogg/vorbis/doc/v-comment.html . This packet has `packetno=1` and `granulepos=0`.

The third and subsequant packets each contain one or more (number found in header) Speex frames. These are identified with `packetno` starting from 2

[1]The wideband bit-stream contains an embedded narrowband bit-stream which can be decoded alone

and the `granulepos` is the number of the last sample encoded in that packet. Le last of these packets has the *end of stream* (`e_o_s`) flag is set to 1.

Field	Type	Size
speex_string	char[]	8
speex_version	char[]	20
speex_header_version	int	4
header_size	int	4
rate	int	4
mode	int	4
mode_bitstream_version	int	4
nb_channels	int	4
bitrate	int	4
frame_size	int	4
vbr	int	4
frames_per_packet	int	4
reserved1	int	4
reserved2	int	4
reserved3	int	4

Table 5: Ogg/Speex header packet

A FAQ

Vorbis is open-source and patent-free, why do we need Speex?

Vorbis is a great project but its goals are not the same as Speex. Vorbis is mostly aimed at compressing music and audio in general, while Speex targets speech only. For that reason Speex can achieve much better results than Vorbis on speech, typically 2-4 times higher compression at equal quality.

Ogg, Speex, Vorbis, what's the difference?

Ogg is a "file format" for holding multimedia data. Vorbis is an audio codec that uses Ogg to store its bit-streams as files, hence the name Ogg Vorbis. Speex also uses the Ogg format to store its bit-streams as files, so technically they would be "Ogg Speex" files (I prefer to call them just Speex files). One difference with Vorbis however, is that Speex is less tied with Ogg. Actually, if what you do is Voice of IP (VoIP), you don't need Ogg at all.

Can I use Speex for compressing music?

Just like Vorbis is not really adapted to speech, Speex is really not adapted for music. In most cases, you'll be better of with Vorbis when it comes to music.

I converted some MP3's to Speex and the quality is bad. What's wrong?

This is called transcoding and it will always result in much poorer quality than the original MP3. Unless you have a really good (size) reason to do so, never transcode speech. This is even valid for self transcoding (tandeming), i.e. If you decode a Speex file and re-encode it again at the same bit-rate, you will lose quality.

Does Speex run on Windows?

As of 0.8.0, Speex can now compile on Windows, though limited testing has been done so far.

Why is encoding so slow compared to decoding?

For most kinds of compression, encoding is inherently slower than decoding. In the case of Speex, encoding consists of finding, for each vector of 5 to 10 samples, the entry that matches the best within a codebook consisting of 16 to 256 entries. On the other hand, at decoding all that needs to be done is lookup the right entry in the codebook using the encoded index. Since a lookup is much faster than a search, the decoder works much faster than the encoder.

Why is Speex so slow on my iPaq (or insert any platform without an FPU)?

Well the parenthesis provides the answer: no FPU (floating-point unit). The Speex code makes heavy use of floating-point operations. On devices with no FPU, all floating-point instructions need to be emulated. This is a very time consuming operation.

I'm getting unusual background noise (hiss) when using libspeex in my application. How do I fix that?

One of the cause could be scaling of the input speech. Speex expects signals to have a $\pm 2^{15}$ (signed short) dynamic range. If the dynamic range of your signals is too small (e.g. ± 1.0), you will suffer important quantization noise. A good target is to have a dynamic range around ± 8000 which is large enough, but small enough to make sure there's no clipping when converting back to signed short.

I get very distorted speech when using libspeex in my application. What's wrong?

There are many possible causes for that. One of them is errors in the way the bits are manipulated. Another possible cause is the use of the same encoder or decoder state for more than one audio stream (channel), which produces strange effects with the filter memories. If the input speech has an amplitude close to $\pm 2^{15}$, it is possible that at decoding, the amplitude be a bit higher than that, causing clipping when saving as 16-bit PCM.

Can Speex pass DTMF?

I guess it all depends on the bit-rate used. Though no formal testing has yet been performed, I'd say don't go below the 15 kbps mode if you want DTMF to be transmitted correctly. DTMF at 8 kbps may work but your mileage may vary. Also, make sure you don't use the lowest complexity (see SPEEX_SET_COMPLEXITY or –comp option), as it causes important noise.

Can Speex pass V.9x modem signals correctly?

If I could to that I'd be very rich by now :-)

What is your (Jean-Marc) relationship with the University of Sherbrooke and how does Speex fit into that?

I am currently (2002/08/13) doing a Ph.D. at the University of Sherbrooke in mobile robotics. Although I did my master with the Sherbrooke speech coding group (in speech enhancement, not coding), I am not associated with them anymore. It should **not** be understood that they or the University of Sherbrooke endorse the Speex project in any way. Furthermore, Speex does not make use of any code or proprietary technology developed in the Sherbrooke speech coding group.

CELP, ACELP, what's the difference?

CELP stands for "Code Excited Linear Prediction", while ACELP stands for "*Algebraic* Code Excited Linear Prediction". That means ACELP is a CELP technique that uses an algebraic codebook represented as a sum of unit pulses, thus making the codebook search much more efficient. This technique was invented at the University of Sherbrooke and is now one of the most widely used form of CELP. Unfortunately, since it is patented, it cannot be used in Speex.

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