Updates to the Opus Audio Codec

Abstract

This document addresses minor issues that were found in the specification of the Opus audio codec in RFC 6716. It updates the normative decoder implementation included in Appendix A of RFC 6716. The changes fix real and potential security-related issues, as well as minor quality-related issues.

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This is an Internet Standards Track document.

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1. Introduction

This document addresses minor issues that were discovered in the reference implementation of the Opus codec. Unlike most IETF specifications, RFC 6716 [RFC6716] defines Opus in terms of a normative reference decoder implementation rather than from the associated text description. Appendix A of that RFC includes the reference decoder implementation, which is why only issues affecting the decoder are listed here. An up-to-date implementation of the Opus encoder can be found at <https://opus-codec.org/>.

Some of the changes in this document update normative behavior in a way that requires new test vectors. Only the C implementation is affected, not the English text of the specification. This specification remains fully compatible with RFC 6716 [RFC6716].

Note: Due to RFC formatting conventions, lines exceeding the column width in the patch are split using a backslash character. The backslashes at the end of a line and the white space at the beginning of the following line are not part of the patch. Referenced line numbers are approximations. A properly formatted patch including all changes is available at <https://www.ietf.org/proceedings/98/slides/materials-98-codec-opus-update-00.patch> and has a SHA-1 hash of 029e3aa88fc342c91e67a21e7fb8c94586616cd5f.
2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Stereo State Reset in SILK

The reference implementation does not reinitialize the stereo state during a mode switch. The old stereo memory can produce a brief impulse (i.e., single sample) in the decoded audio. This can be fixed by changing silk/dec_API.c around line 72:

<CODE BEGINS>
for( n = 0; n < DECODER_NUM_CHANNELS; n++ ) {
   ret  = silk_init_decoder( &channel_state[ n ] );
}
+    silk_memset(&((silk_decoder *)decState)->sStereo, 0,
+                sizeof(((silk_decoder *)decState)->sStereo));
+    /* Not strictly needed, but it’s cleaner that way */
+    ((silk_decoder *)decState)->prev_decode_only_middle = 0;

   return ret;
}
<CODE ENDS>

This change affects the normative output of the decoder, but the amount of change is within the tolerance and is too small to make the test vector check fail.
4. Parsing of the Opus Packet Padding

It was discovered that some invalid packets of a very large size could trigger an out-of-bounds read in the Opus packet parsing code responsible for padding. This is due to an integer overflow if the signaled padding exceeds \(2^{31}-1\) bytes (the actual packet may be smaller). The code can be fixed by decrementing the (signed) `len` value, instead of incrementing a separate padding counter. This is done by applying the following changes around line 596 of `src/opus_decoder.c`:

```
/* Padding flag is bit 6 */
if (ch&0x40)
{
    int padding=0;
    int p;
    do {
        if (len<=0)
            return OPUS_INVALID_PACKET;
        p = *data++;
        len--;
        padding += p==255 ? 254: p;
        len -= p==255 ? 254: p;
    } while (p==255);
    len -= padding;
}
```

This packet-parsing issue is limited to reading memory up to about 60 KB beyond the compressed buffer. This can only be triggered by a compressed packet more than about 16 MB long, so it’s not a problem for RTP. In theory, it could crash a file decoder (e.g., Opus in Ogg) if the memory just after the incoming packet is out of range, but our attempts to trigger such a crash in a production application built using an affected version of the Opus decoder failed.

5. Resampler Buffer

The SILK resampler had the following issues:

1. The calls to `memcpy()` were using `sizeof(opus_int32)`, but the type of the local buffer was `opus_int16`. 
2. Because the size was wrong, this potentially allowed the source and destination regions of the memcpy() to overlap on the copy from "buf" to "buf". We believe that nSamplesIn (number of input samples) is at least fs_in_khZ (sampling rate in kHz), which is at least 8. Since RESAMPLER_ORDER_FIR_12 is only 8, that should not be a problem once the type size is fixed.

3. The size of the buffer used RESAMPLER_MAX_BATCH_SIZE_IN, but the data stored in it was actually twice the input batch size (nSamplesIn<<1).

The code can be fixed by applying the following changes around line 78 of silk/resampler_private_IIR_FIR.c:

<CODE BEGINS>

    silk_resampler_state_struct *S = (silk_resampler_state_struct *)SS;
    opus_int32 nSamplesIn;
    opus_int32 max_index_Q16, index_increment_Q16;
    - opus_int16 buf[ RESAMPLER_MAX_BATCH_SIZE_IN + \n      RESAMPLER_ORDER_FIR_12 ];
    + opus_int16 buf[ 2*RESAMPLER_MAX_BATCH_SIZE_IN + \n      RESAMPLER_ORDER_FIR_12 ];

    /* Copy buffered samples to start of buffer */
    - silk_memcpy( buf, S->sFIR, RESAMPLER_ORDER_FIR_12 \n      * sizeof( opus_int32 ) );
    + silk_memcpy( buf, S->sFIR, RESAMPLER_ORDER_FIR_12 \n      * sizeof( opus_int16 ) );

    /* Iterate over blocks of frameSizeIn input samples */
    index_increment_Q16 = S->invRatio_Q16;
    while( 1 ) {
        nSamplesIn = silk_min( inLen, S->batchSize );

        /* Upsample 2x */
        silk_resampler_private_up2_HQ( S->sIIR, &buf[ \n          RESAMPLER_ORDER_FIR_12 ], in, nSamplesIn );

        max_index_Q16 = silk_LSHIFT32( nSamplesIn, 16 + 1 \n        );
        /* + 1 because 2x upsampling */
        out = silk_resampler_private_IIR_FIR_INTERPOL( out, \n          buf, max_index_Q16, index_increment_Q16 );
        in += nSamplesIn;
        inLen -= nSamplesIn;

<CODE ENDS>
if( inLen > 0 ) {
    /* More iterations to do; copy last part of \n    filtered signal to beginning of buffer */
    silk_memcpy( buf, &buf[nSamplesIn << 1], \n    RESAMPLER_ORDER_FIR_12 * sizeof( opus_int32 ) );
    silk_memmove( buf, &buf[nSamplesIn << 1], \n    RESAMPLER_ORDER_FIR_12 * sizeof( opus_int16 ) );
} else {
    break;
}

/* Copy last part of filtered signal to the state for \nthe next call */
-    silk_memcpy( S->sFIR, &buf[nSamplesIn << 1], \n    RESAMPLER_ORDER_FIR_12 * sizeof( opus_int32 ) );
+    silk_memcpy( S->sFIR, &buf[nSamplesIn << 1], \n    RESAMPLER_ORDER_FIR_12 * sizeof( opus_int16 ) );
}

<CODE ENDS>

6. Integer Wrap-Around in Inverse Gain Computation

It was discovered through decoder fuzzing that some bitstreams could
produce integer values exceeding 32 bits in
LPC_inverse_pred_gain_QA(), causing a wrap-around. The C standard
considers this behavior as undefined. The following patch around
line 87 of silk/LPC_inv_pred_gain.c detects values that do not fit in
a 32-bit integer and considers the corresponding filters unstable:

<CODE BEGINS>
    /* Update AR coefficient */
    for( n = 0; n < k; n++ ) {
        -            tmp_QA = Aold_QA[ n ] - MUL32_FRAC_Q( \n        Aold_QA[ k - n - 1 ], rc_Q31, 31 );
        -            Anew_QA[ n ] = MUL32_FRAC_Q( tmp_QA, rc_mult2 , mult2Q );
        +            opus_int64 tmp64;
        +            tmp_QA = silk_SUB_SAT32( Aold_QA[ n ], MUL32_FRAC_Q( \n        Aold_QA[ k - n - 1 ], rc_Q31, 31 ) );
        +            tmp64 = silk_RSHIFT_ROUND64( silk_SMULL( tmp_QA, \n        rc_mult2 ), mult2Q);
        +            if( tmp64 > silk_int32_MAX || tmp64 < silk_int32_MIN ) { 
        +                return 0;
        +            }
        +            Anew_QA[ n ] = ( opus_int32 )tmp64;
    }
<CODE ENDS>
7. Integer Wrap-Around in LSF Decoding

It was discovered -- also from decoder fuzzing -- that an integer wrap-around could occur when decoding bitstreams with extremely large values for the high Line Spectral Frequency (LSF) parameters. The end result of the wrap-around is an illegal read access on the stack, which the authors do not believe is exploitable but should nonetheless be fixed. The following patch around line 137 of silk/NLSF_stabilize.c prevents the problem:

<CODE BEGINS>
/* Keep delta_min distance between the NLSFs */
for( i = 1; i < L; i++ )
    NLSF_Q15[i] = silk_max_int( NLSF_Q15[i], \n    NLSF_Q15[i-1] + NDeltaMin_Q15[i] );
+    NLSF_Q15[i] = silk_max_int( NLSF_Q15[i], \n    silk_ADD_SAT16( NLSF_Q15[i-1], NDeltaMin_Q15[i] ) );

    /* Last NLSF should be no higher than 1 - NDeltaMin[L] */
<CODE ENDS>

8. Cap on Band Energy

On extreme bitstreams, it is possible for log-domain band energy levels to exceed the maximum single-precision floating point value once converted to a linear scale. This would later cause the decoded values to be NaN (not a number), possibly causing problems in the software using the PCM values. This can be avoided with the following patch around line 552 of celt/quant_bands.c:

<CODE BEGINS>
{ 
    opus_val16 lg = ADD16(oldEBands[i+c*m->nbEBands],
        SHL16((opus_val16)eMeans[i],6));
    +   lg = MIN32(QCONST32(32.f, 16), lg);
    eBands[i+c*m->nbEBands] = PSHR32(celt_exp2(lg),4);
} 
for (;i<m->nbEBands;i++)
<CODE ENDS>
9. Hybrid Folding

When encoding in hybrid mode at low bitrate, we sometimes only have enough bits to code a single Constrained-Energy Lapped Transform (CELT) band (8 - 9.6 kHz). When that happens, the second band (CELT band 18, from 9.6 - 12 kHz) cannot use folding because it is wider than the amount already coded and falls back to white noise. Because it can also happen on transients (e.g., stops), it can cause audible pre-echo.

To address the issue, we change the folding behavior so that it is never forced to fall back to Linear Congruential Generator (LCG) due to the first band not containing enough coefficients to fold onto the second band. This is achieved by simply repeating part of the first band in the folding of the second band. This changes the code in celt/bands.c around line 1237:

```c
b = 0;
}

- if (resynth && M*eBands[i]-N >= M*eBands[start] && 
  (update_lowband || lowband_offset==0))
+ if (resynth && (M*eBands[i]-N >= M*eBands[start] || 
  i==start+1) && (update_lowband || lowband_offset==0))
    lowband_offset = i;
+
+ if (i == start+1)
+ {
+   int n1, n2;
+   int offset;
+   n1 = M*(eBands[start+1]-eBands[start]);
+   n2 = M*(eBands[start+2]-eBands[start+1]);
+   offset = M*eBands[start];
+   /* Duplicate enough of the first band folding data to be able to fold the second band. */
+   OPUS_COPY(&norm[offset+n1], &norm[offset+2*n1 - n2], n2-n1);
+   if (C==2)
+      OPUS_COPY(&norm2[offset+n1], &norm2[offset+2*n1 - n2], n2-n1);
+   }
+
+ tf_change = tf_res[i];
+ if (i>=m->effEBands)
+ {
+ `````
as well as around line 1260:

```
<CODE BEGINS>
fold_start = lowband_offset;
    while(M*eBands[--fold_start] > effective_lowband);
fold_end = lowband_offset-1;
-    while(M*eBands[++fold_end] < effective_lowband+N);
+    while(++fold_end < i && M*eBands[fold_end] < effective_lowband+N);
    effective_lowband+N);
x_cm = y_cm = 0;
fold_i = fold_start; do {
    x_cm |= collapse_masks[fold_i*C+0];
</CODE ENDS>

The fix does not impact compatibility, because the improvement does not depend on the encoder doing anything special. There is also no reasonable way for an encoder to use the original behavior to improve quality over the proposed change.

10. Downmix to Mono

The last issue is not strictly a bug, but it is an issue that has been reported when downmixing an Opus decoded stream to mono, whether this is done inside the decoder or as a post-processing step on the stereo decoder output. Opus intensity stereo allows optionally coding the two channels 180 degrees out of phase on a per-band basis. This provides better stereo quality than forcing the two channels to be in phase, but when the output is downmixed to mono, the energy in the affected bands is canceled, sometimes resulting in audible artifacts.

As a work-around for this issue, the decoder MAY choose not to apply the 180-degree phase shift. This can be useful when downmixing to mono inside or outside of the decoder (e.g., requested explicitly from an API).

11. New Test Vectors

Changes in Sections 9 and 10 have sufficient impact on the test vectors to make them fail. For this reason, this document also updates the Opus test vectors. The new test vectors now include two decoded outputs for the same bitstream. The outputs with suffix ‘m’ do not apply the CELT 180-degree phase shift as allowed in Section 10, while the outputs without the suffix do. An implementation is compliant as long as it passes either set of vectors.
Any Opus implementation that passes either the original test vectors from RFC 6716 [RFC6716] or one of the new sets of test vectors is compliant with the Opus specification. However, newer implementations SHOULD be based on the new test vectors rather than the old ones.

The new test vectors are located at <https://www.ietf.org/proceedings/98/slides/materials-98-codec-opus-newvectors-00.tar.gz>. The SHA-1 hashes of the test vectors are:

```
e49b2826ceec7324790ed8019eb9744596d5be01  testvector01.bit
b809795ae1b2d6060497d6e4ad24236257135e0  testvector02.bit
e04c0caebad4f3d52f5b5675dc996848e5ee2ac  testvector03.bit
a0f870cb8e14b7f1a9066ef3ee9e59ca75187  testvector04.bit
9b3d92b2b48b965dfe9ed7b8a85edd4309f8cf7c  testvector05.bit
28e6676a1b7e17f72875283c14b19690c4ce57  testvector06.bit
bcacf467be3215fc7e2c888f29e2477de1192947a6  testvector07.bit
ddbb0b688b6ef394071f3f893cd0030ce48dab12f  testvector08.bit
3932d961944dab12016458eeead595d5705eb  testvector09.bit
6bce8f314f6cb96450c91b163d3464c7df4d5d96  testvector10.bit
338c3f1fb497226b6c60bc41038bebc56de06828f  testvector12.bit
f5ef93884da6814d311027918e9af6f2e5c2c8  testvector01.dec
48ac1ff199525a0756e1e17bd32acef8ac8dc820  testvector02.dec
d15567e919db2d0e818727092c0aff8dd9df23c95  testvector03.dec
1249dd28a5bd1e39a6f6d699449dca7a8316342  testvector04.dec
b85675081deef84a112c2c66dc7f3b72a1d2f7c6  testvector05.dec
55f0b191e90bfa6f9f8b50d1a6b44255cb4813e  testvector06.dec
61e8b357ab909b1801ebeb578a28a6ae935e2b7b  testvector07.dec
a58539ee5e321453bd2df4c0f2500e856b3966862  testvector08.dec
bb96aad2cde1885558627bb3af6133851ef8f4  testvector09.dec
1b6c6f0413ac9965b16184b1bea129b50b2a37a  testvector10.dec
b1ff7274666e3027801b29d3e483b1f80dee0d  testvector11.dec
98e09bba4f6329e341c3b4052e9c4ba5f83f9b1  testvector12.dec
1e79d84e3afbb16a998ae761f4893fbd30157  testvector01m.dec
48ac1ff199525a0756e1e17bd32acef8ac8dc820  testvector02m.dec
d15567e919db2d0e818727092c0aff8dd9df23c95  testvector03m.dec
1249dd28a5bd1e39a6f6d699449dca7a8316342  testvector04m.dec
70bfbbad4317e7d463bc3da49bd2d49f1c60a530  testvector05m.dec
6ac1648c3174c95a655161a6c78dbde59c577d  testvector06m.dec
fc5e2f70969373824fb4c8bdc0dad6dada04e713  testvector07m.dec
aadd2b937bbf16a18e8e09b50e4b19627d479f00  testvector08m.dec
6fe7a7b9d7cc1383babf8d57392a514bd0b0a8  testvector09m.dec
1b6c6f0413ac9965b16184b1bea129b50b2a37a  testvector10m.dec
fd3d3a7b0dfdbab98d37ed9aa04b659b9febd18  testvector11m.dec
98e09bba4f6329e341c3b4052e9c4ba5f83f9b1  testvector12m.dec
```

Note that the decoder input bitstream files (.bit) are unchanged.
12. Security Considerations

This document fixes two security issues reported on Opus that affect the reference implementation in RFC 6716 [RFC6716]: CVE-2013-0899 <https://nvd.nist.gov/vuln/detail/CVE-2013-0899> and CVE-2017-0381 <https://nvd.nist.gov/vuln/detail/CVE-2017-0381>. CVE-2013-0899 theoretically could have caused an information leak. The leaked information would have gone through the decoder process before being accessible to the attacker. The update in Section 4 fixes this.

CVE-2017-0381 could have resulted in a 16-bit out-of-bounds read from a fixed location. The update in Section 7 fixes this. Beyond the two fixed Common Vulnerabilities and Exposures (CVEs), this document adds no new security considerations beyond those in RFC 6716 [RFC6716].

13. IANA Considerations

This document does not require any IANA actions.

14. Normative References


Acknowledgements

We would like to thank Juri Aedla for reporting the issue with the parsing of the Opus padding. Thanks to Felicia Lim for reporting the LSF integer overflow issue. Also, thanks to Tina le Grand, Jonathan Lennox, and Mark Harris for their feedback on this document.
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