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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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 029e3aa88fc342c91e67a21e7bfbc94586d5f5f.
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:

```c
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;
}
```

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:

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

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

        max_index_Q16 = silk_LSHIFT32( nSamplesIn, 16 + 1 \
          );
        /* + 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 \ filtered signal to beginning of buffer */
    silk_memcpy( buf, &buf[ nSamplesIn << 1 ], \
                RESAMPLER_ORDER_FIR_12 * sizeof( opus_int32 ) );
    +            silk_memmove( buf, &buf[ nSamplesIn << 1 ], \
                        RESAMPLER_ORDER_FIR_12 * sizeof( opus_int16 ) );
    } else {
        break;
    }
}

/* Copy last part of filtered signal to the state for \ the next call */
-    silk_memcpy( S->sFIR, &buf[ nSamplesIn << 1 ], \
                 RESAMPLER_ORDER_FIR_12 * sizeof( opus_int32 ) );
+    silk_memcpy( S->sFIR, &buf[ nSamplesIn << 1 ], \
                 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(  
               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(  
                   Aold_QA[ k - n - 1 ], rc_Q31, 31 ) );
+            tmp64 = silk_RSHIFT_ROUND64( silk_SMULL( tmp_QA,  
                               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:

```c
/* 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] */
```

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:

```c
{  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++)
```

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)
    {
```

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as well as around line 1260:

```c
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] < \n  effective_lowband+N);
x_cm = y_cm = 0;
fold_i = fold_start; do {
  x_cm |= collapse_masks[fold_i*C+0];
```

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:

e49b2862c1ee73724790ed8019e9b7944596d5be01  testvector01.bit
b809795ae1bced606049d76de4ad24236257135e0  testvector02.bit
e0c4eaeab44d35a2f5b657cd996848e5ee2acc  testvector03.bit
a0f870cbe14ebb771fa9066f3ee96e59c9a75187  testvector04.bit
9b3d92b48b965dfef9edef7b8a16d4309f8cf7c8  testvector05.bit
28e66769a1b1e17f72875283c14b19690c4e57  testvector06/bit
bacf467be3215fc7ec288f29e2477de1192947a6  testvector07.bit
dde08b688bef934071f3893cd0030ce48db12f  testvector08.bit
3932d9d61944d1ab101645b8eaaad595d5705ecb  testvector09.bit
521eb2a1b3c93c71b87407673307c23d3b10c1900 testvector10.bit
6bc83146fcb8645b46633d4c64c8d4d596  testvector11.bit
338cf1b497226bc60bc41038bebc6de06b28f testvector12.bit
f5e93884da6a814d311027918e9af6f2e5c2c8  testvector01.dec
48ac1ff1995250a756e1e17bd32acefa8cd2b820 testvector02.dec
d15567e199b2d038e18727092c0af8dd9df23c95 testvector03.dec
1249dd28f5bd1e39a66fd699449dca7a8316342 testvector04.dec
b85675b81deef84a112c466c6df3b7aaa1d2fc76 testvector05.dec
55f0b19e01baf6f98b50d1a65b44255cb4813e testvector06.dec
61e8357ab0b09b1801ebb578a28a6a93e5257b testvector07.dec
a58539ee532145b2ddf4c0f2500e856b396862 testvector08.dec
bb9a6ad2c2eb8855582b7b8bb3af613851ef8f4 testvector09.dec
1b6c0f4013ac9965b16164b1bea129b5c0a2a37a testvector10.dec
blff27b7466e3027801b29db4831f80de0d testvector11.dec
98e09bbafed329e341c3b4052e9c4ba5f83f9b1 testvector12.dec
1e7d989ea3fbb1a6ba998ae761f4893fbdb30157 testvector01m.dec
48ac1ff1995250a756e1e17bd32acefa8cd2b820 testvector02m.dec
d15567e199b2d038e18727092c0af8dd9df23c95 testvector03m.dec
1249dd28f5bd1e39a66fd699449dca7a8316342 testvector04m.dec
b85675b81deef84a112c466c6df3b7aaa1d2fc76 testvector05m.dec
55f0b19e01baf6f98b50d1a65b44255cb4813e testvector06m.dec
61e8357ab0b09b1801ebb578a28a6a93e5257b testvector07m.dec
a58539ee532145b2ddf4c0f2500e856b396862 testvector08m.dec
bb9a6ad2c2eb8855582b7b8bb3af613851ef8f4 testvector09m.dec
1b6c0f4013ac9965b16164b1bea129b5c0a2a37a testvector10m.dec
blff27b7466e3027801b29db4831f80de0d testvector11m.dec
98e09bbafed329e341c3b4052e9c4ba5f83f9b1 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


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Authors’ Addresses

Jean-Marc Valin
Mozilla Corporation
331 E. Evelyn Avenue
Mountain View, CA  94041
United States of America

Phone: +1 650 903-0800
Email: jmvalin@jmvalin.ca

Koen Vos
vocTone

Email: koenvos74@gmail.com