

Network Working Group
Internet-Draft
Intended status: Standards Track
Expires: March 9, 2015

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September 5, 2014

Updates to the Opus Audio Codec
draft-ietf-codec-opus-update-01

Abstract

This document addresses minor issues that were found in the specification of the Opus audio codec in [RFC 6716](#) [[RFC6716](#)].

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[1.](#) Introduction

This document addresses minor issues that were discovered in the reference implementation of the Opus codec that serves as the specification in [RFC 6716](#) [[RFC6716](#)]. Only issues affecting the decoder are listed here. An up-to-date implementation of the Opus encoder can be found at <http://opus-codec.org/>. The updated specification remains fully compatible with the original specification and only one of the changes results in any difference in the audio output.

[2.](#) Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC 2119](#) [[RFC2119](#)].

[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` at line 72:

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```
    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 part of the decoder. Fortunately, the modified decoder is still compliant with the original specification because it still easily passes the testvectors. For example, for the float decoder at 48 kHz, the `opus_compare` (arbitrary) "quality score" changes from from 99.9333% to 99.925%.

[4.](#) Parsing of the Opus Packet Padding

It was discovered that some invalid packets of 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 applying the following changes at 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. We believe that `nSamplesIn` is at least `fs_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 fact that the code never produced any error in testing (including when run under the Valgrind memory debugger), suggests that in practice the batch sizes are reasonable enough that none of the issues above was ever a problem. However, proving that is non-obvious.

The code can be fixed by applying the following changes to line 70 of `silk/resampler_private_IIR_FIR.c`:

```

)
{
    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 + \
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, \
buf, max_index_Q16, index_increment_Q16 );
    in += nSamplesIn;
    inLen -= nSamplesIn;

    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 ) );
}

```

Note: due to RFC formatting conventions, lines exceeding the column width in the patch above 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. A properly formatted patch including the three changes above is available at [\[1\]](#).

[6.](#) 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 cancelled sometimes resulting in audible artefacts.

As a work-around for this issue, the decoder MAY choose not to apply the 180-degree phase shift when the output is meant to be downmixed (inside or outside of the decoder).

[7.](#) IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section may be removed on publication as an RFC.

8. Acknowledgements

We would like to thank Juri Aedla for reporting the issue with the parsing of the Opus padding.

9. References

9.1. References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", [BCP 14](#), [RFC 2119](#), March 1997.

[RFC6716] Valin, JM., Vos, K., and T. Terriberry, "Definition of the Opus Audio Codec", [RFC 6716](#), September 2012.

9.2. URIs

[1] http://jmvalin.ca/misc_stuff/opus_update.patch

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