avtcore WG
Internet-Draft
Intended status: Informational
Expires: 22 April 2026

S. Shin
NVIDIA

19 October 2025

RTP/SDP for Opus Multistream
draft-shin-avtcore-rtp-multi-opus-01

Abstract

This document specifies RTP/SDP signaling for Opus multistream (multi-channel) operation, enabling negotiation of layouts such as 5.1 and 7.1 in real-time communications. It defines an SDP encoding name and format parameters to describe multistream configurations, and specifies Offer/Answer procedures for interoperable negotiation. This document extends the Opus RTP payload format defined in [RFC7587] and reuses the channel-mapping concepts defined for the Ogg container in [RFC7845].

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at https://datatracker.ietf.org/drafts/current/.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on 22 April 2026.

Copyright Notice

Table of Contents

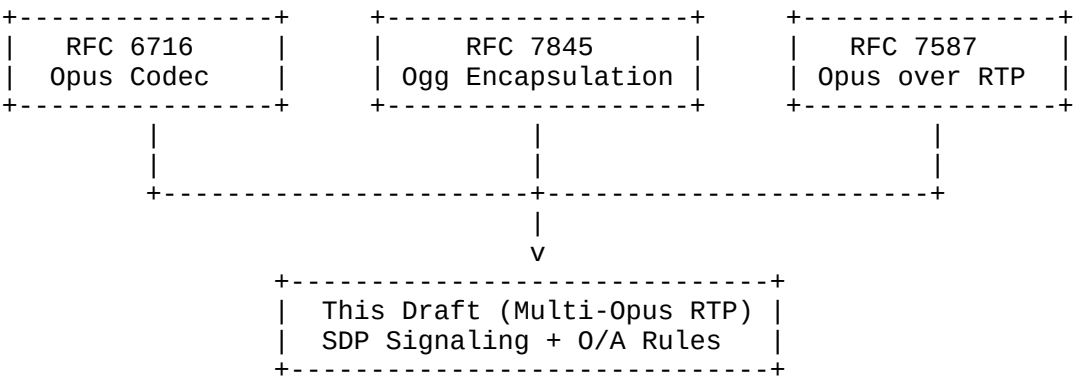
1. Introduction

Opus ([RFC6716]) supports up to 255 channels via multistream with explicit channel mapping. The RTP payload format for Opus ([RFC7587]), however, standardizes only mono/stereo signaling for RTP/SDP and leaves multistream out of scope. [RFC7845] defines channel-mapping families for Opus carried in the Ogg container, but it does not define RTP/SDP signaling or Offer/Answer behavior. This document fills that gap by specifying interoperable SDP signaling and Offer/Answer procedures for multistream Opus in RTP sessions, while aligning channel-mapping semantics with [RFC7845].

2. Relationship to Existing RFCs

This section summarizes the scope and relationship between [RFC6716] (Opus codec), [RFC7587] (Opus over RTP), [RFC7845] (Ogg encapsulation of Opus), and this draft. While RFC 7845 defines channel mapping families for multistream Opus in the Ogg container, it does not define SDP signaling or RTP usage. [RFC7587] defines the RTP payload format for Opus but only covers mono/stereo signaling. This draft extends [RFC7587] to define SDP signaling for multistream Opus in RTP sessions and reuses the mapping semantics from [RFC7845].

3. Relationship of RFCs and This Draft:



4. Summary of RFCs and This Draft:

RFC/Draft	Scope	Defines Channel Map?	Defines SDP Signaling
RFC 6716	Opus codec	Yes (API level)	No
RFC 7845	Ogg encapsulation	Yes (families)	No
RFC 7587	RTP payload format	No (mono/stereo only)	Yes (mono/stereo)
This Draft	RTP multistream SDP	Reuses RFC 7845	Yes (multi-channel)

5. Overview and Rationale

Deployed systems (e.g., [libwebrtc] based) interoperate using a non-standard SDP encoding name “multiopus” with fmtp parameters such as num_streams, coupled_streams, and channel_mapping. Standardizing these semantics improves interoperability and removes the need for application-level SDP text modifications.

6. SDP Signaling for Opus Multistream

6.1. SDP Syntax for Multichannel Opus

6.1.1. SDP Example for 5.1 Audio

```
sdp a=rtpmap:111 multiopus/48000/6 a=fmtp:111
num_streams=4;coupled_streams=2;channel_mapping=0,4,1,2,3,5
```

6.1.2. SDP Example for 7.1 Audio

```
sdp a=rtpmap:111 multiopus/48000/8 a=fmtp:111
num_streams=5;coupled_streams=3;channel_mapping=0,6,1,2,3,4,5,7
```

6.1.3. Field Descriptions

* a=rtpmap:<pt> multiopus/48000/<channel-count>
  - <pt>: Dynamic payload type (e.g., 96).
  - 48000: Fixed clock rate for Opus.
  - <channel-count>: Total number of output channels (e.g., 6 for 5.1, 8 for 7.1).
* a=fmtp:<pt> num_streams=<N>;coupled_streams=<M>;channel_mapping=<C>
  - num_streams: Total number of Opus streams.
  - coupled_streams: Number of stereo (coupled) streams.
  - channel_mapping: Comma-separated list mapping RTP channels to speaker positions.
* The channel_mapping values follow the Opus multistream mapping used by [libwebrtc].

7. Offer/Answer Procedures

An offerer willing to negotiate multichannel Opus MAY include one or more payload types using multiopus with appropriate fmtp, and SHOULD include a stereo alternative using opus/48000/2 ([RFC7587]) for backward compatibility.

An answerer that supports the offered multiopus configuration MUST select the corresponding payload type and include the selected multistream parameters in the answer.

If unsupported, the answerer MAY select a stereo opus payload or reject the m-section per [RFC3264]. Down-conversion to stereo SHOULD NOT occur silently when the answerer supports the offered configuration.

7.1. Examples

7.1.1. Offer: 5.1 (6 channels)

```
m=audio 9 UDP/TLS/RTP/SAVPF 111 112 a=mid:audio a=rtpmap:111
multiopus/48000/6 a=fmtp:111
num_streams=4;coupled_streams=2;channel_mapping=0,4,1,2,3,5
a=rtpmap:112 opus/48000/2 a=sendrecv
```

7.1.2. Answer: accept 5.1

```
m=audio 9 UDP/TLS/RTP/SAVPF 111 a=mid:audio a=rtpmap:111
multiopus/48000/6 a=fmtp:111
num_streams=4;coupled_streams=2;channel_mapping=0,4,1,2,3,5
a=sendrecv
```

7.2. Security Considerations

The use of the multiopus encoding in SDP does not introduce new security concerns beyond those already described in [RFC7587]. Implementers should ensure that SDP parsing and RTP payload handling are robust against malformed or malicious input. Applications using multichannel Opus streams must also consider the privacy implications of transmitting spatial audio data, which may reveal environmental context.

Transport-level security mechanisms such as DTLS-SRTP are recommended to protect RTP streams.

7.3. IANA Considerations

This document does not require any new IANA registrations. The multiopus encoding name and associated SDP attributes are used in accordance with existing conventions and do not introduce new protocol elements.

8. Informative References

[libwebrtc]
WebRTC, "Opus multistream mapping", n.d., <https://webrtc-review.googlesource.com/c/src/+/129768>.

[RFC3264] Rosenberg, J. and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)", RFC 3264, DOI 10.17487/RFC3264, June 2002, <https://www.rfc-editor.org/rfc/rfc3264>.

[RFC6716] Valin, JM., Vos, K., and T. Terriberry, "Definition of the Opus Audio Codec", RFC 6716, DOI 10.17487/RFC6716, September 2012, <https://www.rfc-editor.org/rfc/rfc6716>.

[RFC7587] Spittka, J., Vos, K., and JM. Valin, "RTP Payload Format for the Opus Speech and Audio Codec", RFC 7587, DOI 10.17487/RFC7587, June 2015, <https://www.rfc-editor.org/rfc/rfc7587>.

[RFC7845] Terriberry, T., Lee, R., and R. Giles, "Ogg Encapsulation for the Opus Audio Codec", RFC 7845, DOI 10.17487/RFC7845, April 2016, <https://www.rfc-editor.org/rfc/rfc7845>.

Author’s Address

Sun Shin
NVIDIA
Email: sushin@nvidia.com