

**Framing Real-time Transport Protocol (RTP)
and RTP Control Protocol (RTCP) Packets
over Connection-Oriented Transport**

Status of This Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

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Abstract

This memo defines a method for framing Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP) packets onto connection-oriented transport (such as TCP). The memo also defines how session descriptions may specify RTP streams that use the framing method.

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1. Introduction

The Audio/Video Profile (AVP, [RFC3550]) for the Real-time Transport Protocol (RTP, [RFC3551]) does not define a method for framing RTP and RTP Control Protocol (RTCP) packets onto connection-oriented transport protocols (such as TCP). However, earlier versions of RTP/AVP did define a framing method, and this method is in use in several implementations.

In this memo, we document the framing method that was defined by earlier versions of RTP/AVP. In addition, we introduce a mechanism for a session description [SDP] to signal the use of the framing method. Note that session description signalling for the framing method is new and was not defined in earlier versions of RTP/AVP.

1.1. 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 BCP 14, RFC 2119 [RFC2119].

2. The Framing Method

Figure 1 defines the framing method.

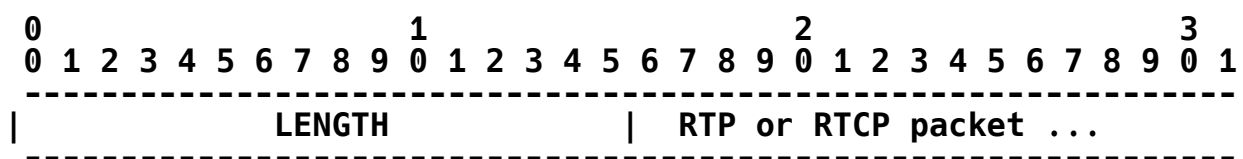


Figure 1: The bit field definition of the framing method

A 16-bit unsigned integer LENGTH field, coded in network byte order (big-endian), begins the frame. If LENGTH is non-zero, an RTP or RTCP packet follows the LENGTH field. The value coded in the LENGTH field MUST equal the number of octets in the RTP or RTCP packet. Zero is a valid value for LENGTH, and it codes the null packet.

This framing method does not use frame markers (i.e., an octet of constant value that would precede the LENGTH field). Frame markers are useful for detecting errors in the LENGTH field. In lieu of a frame marker, receivers SHOULD monitor the RTP and RTCP header fields whose values are predictable (for example, the RTP version number). See Appendix A.1 of [RFC3550] for additional guidance.

3. Packet Stream Properties

In most respects, the framing method does not specify properties above the level of a single packet. In particular, Section 2 does not specify the following:

Bi-directional issues

Section 2 defines a framing method for use in one direction on a connection. The relationship between framed packets flowing in a defined direction and in the reverse direction is not specified.

Packet loss and reordering

The reliable nature of a connection does not imply that a framed RTP stream has a contiguous sequence number ordering. For example, if the connection is used to tunnel a UDP stream through a network middlebox that only passes TCP, the sequence numbers in the framed stream reflect any packet loss or reordering on the UDP portion of the end-to-end flow.

Out-of-band semantics

Section 2 does not define the RTP or RTCP semantics for closing a TCP socket, or of any other "out of band" signal for the connection.

Memos that normatively include the framing method MAY specify these properties. For example, Section 4 of this memo specifies these properties for RTP/AVP sessions specified in session descriptions.

In one respect, the framing protocol does indeed specify a property above the level of a single packet. If a direction of a connection carries RTP packets, the streams carried in this direction MUST support the use of multiple synchronization sources (SSRCs) in those RTP packets. If a direction of a connection carries RTCP packets, the streams carried in this direction MUST support the use of multiple SSRCs in those RTCP packets.

4. Session Descriptions for RTP/AVP over TCP

Session management protocols that use the Session Description Protocol [SDP] in conjunction with the Offer/Answer Protocol [RFC3264] MUST use the methods described in [COMEDIA] to set up RTP/AVP streams over TCP. In this case, the use of Offer/Answer is REQUIRED, as the setup methods described in [COMEDIA] rely on Offer/Answer.

In principle, [COMEDIA] is capable of setting up RTP sessions for any RTP profile. In practice, each profile has unique issues that must be considered when applying [COMEDIA] to set up streams for the profile.

In this memo, we restrict our focus to the Audio/Video Profile (AVP, [RFC3551]). Below, we define a token value ("TCP/RTP/AVP") that signals the use of RTP/AVP in a TCP session. We also define the operational procedures that a TCP/RTP/AVP stream **MUST** follow.

We expect that other standards-track memos will appear to support the use of the framing method with other RTP profiles. The support memo for a new profile **MUST** define a token value for the profile, using the style we used for AVP. Thus, for profile xyz, the token value **MUST** be "TCP/RTP/xyz". The memo **SHOULD** adopt the operational procedures we define below for AVP, unless these procedures are in some way incompatible with the profile.

The remainder of this section describes how to setup and use an AVP stream in a TCP session. Figure 2 shows the syntax of a media (m=) line [SDP] of a session description:

```
"m=" media SP port ["/" integer] SP proto 1*(SP fmt) CRLF
```

Figure 2: Syntax for an SDP media (m=) line (from [SDP])

The <proto> token value "TCP/RTP/AVP" specifies an RTP/AVP [RFC3550] [RFC3551] stream that uses the framing method over TCP.

The <fmt> tokens that follow <proto> **MUST** be unique unsigned integers in the range 0 to 127. The <fmt> tokens specify an RTP payload type associated with the stream.

In all other respects, the session description syntax for the framing method is identical to [COMEDIA].

The TCP <port> on the media line carries RTP packets. If a media stream uses RTCP, a second connection carries RTCP packets. The port for the RTCP connection is chosen using the algorithms defined in [SDP] or by the mechanism defined in [RFC3605].

The TCP connections **MAY** carry bi-directional traffic, following the semantics defined in [COMEDIA]. Both directions of a connection **MUST** carry the same type of packets (RTP or RTCP). The packets **MUST** exclusively code the RTP or RTCP streams specified on the media line(s) associated with the connection.

As noted in [RFC3550], the use of RTP without RTCP is strongly discouraged. However, if a sender does not wish to send RTCP packets in a media session, the sender **MUST** add the lines "b=RS:0" AND "b=RR:0" to the media description (from [RFC3556]).

If the session descriptions of the offer AND the answer both contain the "b=RS:0" AND "b=RR:0" lines, an RTCP TCP flow for the media session **MUST NOT** be created by either endpoint in the session. In all other cases, endpoints **MUST** establish two TCP connections for an RTP/AVP stream, one for RTP and one for RTCP.

As described in [RFC3264], the use of the "sendonly" or "sendrecv" attribute in an offer (or answer) indicates that the offerer (or answerer) intends to send RTP packets on the RTP TCP connection. The use of the "recvonly" or "sendrecv" attributes in an offer (or answer) indicates that the offerer (or answerer) wishes to receive RTP packets on the RTP TCP connection.

5. Example

The session descriptions in Figures 3 and 4 define a TCP RTP/AVP session.

```
v=0
o=first 2520644554 2838152170 IN IP4 first.example.net
s=Example
t=0 0
c=IN IP4 192.0.2.105
m=audio 9 TCP/RTP/AVP 11
a=setup:active
a=connection:new
```

Figure 3: TCP session description for the first participant

```
v=0
o=second 2520644554 2838152170 IN IP4 second.example.net
s=Example
t=0 0
c=IN IP4 192.0.2.94
m=audio 16112 TCP/RTP/AVP 10 11
a=setup:passive
a=connection:new
```

Figure 4: TCP session description for the second participant

The session descriptions define two parties that participate in a connection-oriented RTP/AVP session. The first party (Figure 3) is capable of receiving stereo L16 streams (static payload type 11).

The second party (Figure 4) is capable of receiving mono (static payload type 10) or stereo L16 streams.

The "setup" attribute in Figure 3 specifies that the first party is "active" and initiates connections, and the "setup" attribute in Figure 4 specifies that the second party is "passive" and accepts connections [COMEDIA].

The first party connects to the network address (192.0.2.94) and port (16112) of the second party. Once the connection is established, it is used bi-directionally: the first party sends framed RTP packets to the second party in one direction of the connection, and the second party sends framed RTP packets to the first party in the other direction of the connection.

The first party also initiates an RTCP TCP connection to port 16113 (16112 + 1, as defined in [SDP]) of the second party. Once the connection is established, the first party sends framed RTCP packets to the second party in one direction of the connection, and the second party sends framed RTCP packets to the first party in the other direction of the connection.

6. Congestion Control

The RTP congestion control requirements are defined in [RFC3550]. As noted in [RFC3550], all transport protocols used on the Internet need to address congestion control in some way, and RTP is not an exception.

In addition, the congestion control requirements for the Audio/Video Profile are defined in [RFC3551]. The basic congestion control requirement defined in [RFC3551] is that RTP sessions should compete fairly with TCP flows that share the network. As the framing method uses TCP, it competes fairly with other TCP flows by definition.

7. Acknowledgements

This memo, in part, documents discussions on the AVT mailing list about TCP and RTP. Thanks to all of the participants in these discussions.

8. Security Considerations

Implementors should carefully read the Security Considerations sections of the RTP [RFC3550] and RTP/AVP [RFC3551] documents, as most of the issues discussed in these sections directly apply to RTP streams framed over TCP.

Session descriptions that specify connection-oriented media sessions (such as the example session shown in Figures 3 and 4 of Section 5) raise unique security concerns for streaming media. The Security Considerations section of [COMEDIA] describes these issues in detail.

Below, we discuss security issues that are unique to the framing method defined in Section 2.

Attackers may send framed packets with large LENGTH values to exploit security holes in applications. For example, a C implementation may declare a 1500-byte array as a stack variable, and use LENGTH as the bound on the loop that reads the framed packet into the array. This code would work fine for friendly applications that use Etherframe-sized RTP packets, but may be open to exploit by an attacker. Thus, an implementation needs to handle packets of any length, from a NULL packet (LENGTH == 0) to the maximum-length packet holding 64K octets (LENGTH = 0xFFFF).

9. IANA Considerations

[SDP] defines the syntax of session description media lines. We reproduce this definition in Figure 2 of Section 4 of this memo. In Section 4, we define a new token value for the <proto> field of media lines: "TCP/RTP/AVP". Section 4 specifies the semantics associated with the <proto> field token, and Section 5 shows an example of its use in a session description.

10. Normative References

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