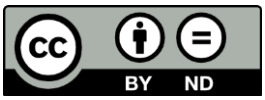
Video Services Forum (VSF)
Technical Recommendation TR-06-1

Reliable Internet Stream Transport (RIST) Protocol Specification – Simple Profile

October 17, 2018

VSF_TR-06_2018_10_17



INTELLECTUAL PROPERTY RIGHTS

THE FORUM DRAWS ATTENTION TO THE FACT THAT IT IS CLAIMED THAT COMPLIANCE WITH THIS RECOMMENDATION MAY INVOLVE THE USE OF A PATENT ("IPR") CONCERNING SECTIONS 4, 5 , 5.1 (EXCLUDING 5.1.2), 5.2 , 5.3 AND ITS SUB SECTIONS, AND 5.4.

THE FORUM TAKES NO POSITION CONCERNING THE EVIDENCE, VALIDITY OR SCOPE OF THIS IPR.

THE HOLDER OF THIS IPR HAS ASSURED THE FORUM THAT IT IS WILLING TO LICENSE ALL IPR IT OWNS AND ANY THIRD PARTY IPR IT HAS THE RIGHT TO SUBLICENSE WHICH MIGHT BE INFRINGED BY ANY IMPLEMENTATION OF THIS RECOMMENDATION TO THE FORUM AND THOSE LICENSEES (MEMBERS AND NON-MEMBERS ALIKE) DESIRING TO IMPLEMENT THIS RECOMMENDATION. INFORMATION MAY BE OBTAINED FROM:

VIDEO-FLOW.LTD
11 HA'AMAL ST. ROSH HA'AYIN ISRAEL, 4809241

ATTENTION IS ALSO DRAWN TO THE POSSIBILITY THAT SOME OF THE ELEMENTS OF THIS RECOMMENDATION MAY BE THE SUBJECT OF IPR OTHER THAN THOSE IDENTIFIED ABOVE. THE FORUM SHALL NOT BE RESPONSIBLE FOR IDENTIFYING ANY OR ALL SUCH IPR.

THIS RECOMMENDATION IS BEING OFFERED WITHOUT ANY WARRANTY WHATSOEVER, AND IN PARTICULAR, ANY WARRANTY OF NONINFRINGEMENT IS EXPRESSLY DISCLAIMED. ANY USE OF THIS RECOMMENDATION SHALL BE MADE ENTIRELY AT THE IMPLEMENTER'S OWN RISK, AND NEITHER THE FORUM, NOR ANY OF ITS MEMBERS OR SUBMITTERS, SHALL HAVE ANY LIABILITY WHATSOEVER TO ANY MPLEMENTER OR THIRD PARTY FOR ANY DAMAGES OF ANY NATURE WHATSOEVER, DIRECTLY OR INDIRECTLY, ARISING FROM THE USE OF THIS RECOMMENDATION.

LIMITATION OF LIABILITY

VSF SHALL NOT BE LIABLE FOR ANY AND ALL DAMAGES, DIRECT OR INDIRECT, ARISING FROM OR RELATING TO ANY USE OF THE CONTENTS CONTAINED HEREIN, INCLUDING WITHOUT LIMITATION ANY AND ALL INDIRECT, SPECIAL, INCIDENTAL OR CONSEQUENTIAL DAMAGES (INCLUDING DAMAGES FOR LOSS OF BUSINESS, LOSS OF PROFITS, LITIGATION, OR THE LIKE), WHETHER BASED UPON BREACH OF CONTRACT, BREACH OF WARRANTY, TORT (INCLUDING NEGLIGENCE), PRODUCT LIABILITY OR OTHERWISE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGES. THE FOREGOING NEGATION OF DAMAGES IS A FUNDAMENTAL ELEMENT OF THE USE OF THE CONTENTS HEREOF, AND THESE CONTENTS WOULD NOT BE PUBLISHED BY VSF WITHOUT SUCH LIMITATIONS.

Executive Summary

Many solutions exist in the market for reliable streaming over the Internet. These solutions all use the same types of techniques, but they are all proprietary and do not interoperate. This Technical Recommendation contains a protocol specification for reliable streaming over the Internet, so end users can mix and match solutions from different vendors.

Recipients of this document are requested to submit notification of any relevant patent claims or other intellectual property rights of which they may be aware that might be infringed by any implementation of the Recommendation set forth in this document, and to provide supporting documentation.

Table of Contents

1 Introduction

As broadcasters and other video users increasingly utilize unconditioned Internet circuits to transport high-quality video, the demand grows for systems that can compensate for the packet losses and delay variation that often affect these streams. A variety of solutions are currently available on the market, however, incompatibilities exist between devices from different suppliers.

The Reliable Internet Stream Transport (RIST) project was launched specifically to address the lack of compatibility between devices, and to define a set of interoperability points through the use of existing or new standards and recommendations.

1.1 Contributors

The following individuals participated in the Video Services Forum RIST working group which developed this technical recommendation.

Merrick Ackermans (MVA Broadcast Consulting)	Sergio M Ammirata (DVEO)	Paul Atwell (Media Transport Solutions)
Uri Avni (Zixi)	John Beer (QVidium)	Ghislain Collette (Haivision)
Magnus Danielson (NetInsight)	Israel Drori (Zixi)	Eric Fankhauser (Evertz)
Ronald Fellman (QVidium)	Michael Firth (Nevion)	Rafael Fonseca (Artel)
Oded Gants (Zixi)	Peter Keys (Charter Communications)	Holger Klaas (Nevion)
Brian Matherly (Sencore)	Ciro Noronha (Cobalt Digital)	Andy Rayner (Nevion)
Steve Riedl (Turner)	David Robison (CenturyLink)	Adi Rozenberg (VideoFlow)
Bob Ruhl (VSF)	Wes Simpson (Telecom Product Consulting)	Adam Yellen (Haivision)

1.2 About the Video Services Forum

The Video Services Forum, Inc. (www.videoservicesforum.org) is an international association dedicated to video transport technologies, interoperability, quality metrics and education. The VSF is composed of service providers, users and manufacturers. The organization's activities include:

- providing forums to identify issues involving the development, engineering, installation, testing and maintenance of audio and video services;
- exchanging non-proprietary information to promote the development of video transport service technology and to foster resolution of issues common to the video services industry;
- identification of video services applications and educational services utilizing video transport services;

- promoting interoperability and encouraging technical standards for national and international standards bodies.

The VSF is an association incorporated under the Not For Profit Corporation Law of the State of New York. Membership is open to businesses, public sector organizations and individuals worldwide. For more information on the Video Services Forum or this document, please call +1 929-279-1995 or e-mail opsmgr@videoservicesforum.org.

2 Conformance Notation

Normative text is text that describes elements of the design that are indispensable or contains the conformance language keywords: "shall", "should", or "may". Informative text is text that is potentially helpful to the user, but not indispensable, and can be removed, changed, or added editorially without affecting interoperability. Informative text does not contain any conformance keywords.

All text in this document is, by default, normative, except the Introduction and any section explicitly labeled as "Informative" or individual paragraphs that start with "Note:"

The keywords "shall" and "shall not" indicate requirements strictly to be followed in order to conform to the document and from which no deviation is permitted.

The keywords, "should" and "should not" indicate that, among several possibilities, one is recommended as particularly suitable, without mentioning or excluding others; or that a certain course of action is preferred but not necessarily required; or that (in the negative form) a certain possibility or course of action is deprecated but not prohibited.

The keywords "may" and "need not" indicate courses of action permissible within the limits of the document.

The keyword “reserved” indicates a provision that is not defined at this time, shall not be used, and may be defined in the future. The keyword “forbidden” indicates “reserved” and in addition indicates that the provision will never be defined in the future.

A conformant implementation according to this document is one that includes all mandatory provisions ("shall") and, if implemented, all recommended provisions ("should") as described. A conformant implementation need not implement optional provisions ("may") and need not implement them as described.

Unless otherwise specified, the order of precedence of the types of normative information in this document shall be as follows: Normative prose shall be the authoritative definition; Tables shall be next; followed by formal languages; then figures; and then any other language forms.

3 References (normative)

SMPTE ST 2022-1:2007, Forward Error Correction for Real-Time Video/Audio Transport Over IP Networks

SMPTE ST 2022-2:2007, Unidirectional Transport of Constant Bit Rate MPEG-2 Transport Streams on IP Networks

SMPTE ST 2022-7:2013, Seamless Protection Switching of SMPTE ST 2022 IP Datagrams

IETF RFC 3550, RTP: A Transport Protocol for Real-Time Applications

IETF RFC 3551, RTP Profile for Audio and Video Conferences with Minimal Control

IETF RFC 4585, Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVFP)

4 RIST Profiles

RIST will have multiple operational profiles, corresponding to increasing levels of complexity and functionality. Higher profiles will include all the features and functionality of the preceding profiles. This document defines RIST Simple Profile. A profile roadmap is included in a companion document.

5 Simple Profile

RIST Profile 1 provides only basic interoperability and packet loss recovery. All configuration is manual and done outside the protocol.

5.1 Baseline Protocol

In order to ensure a level of interoperability between RIST and non-RIST implementations, RTP shall be used as the baseline protocol for media transport. If an RTP standard exists for a certain media type, that standard shall be used as the definition of the RTP header fields. For example, if the media to be transported is in the format of an MPEG-2 Transport Stream, SMPTE-2022-1/2 shall be used for the baseline stream. RIST will augment the baseline RTP transmission with mechanisms to recover from packet loss.

Feedback/control messages shall use RTCP, as specified in RFC 3550.

5.1.1 Unicast Port Assignments

In a unicast environment (such as most of the Internet), RIST senders will transmit to the unicast IP addresses of the RIST receivers. In this environment, the following rules will apply:

1. RIST senders shall transmit the RTP media packets to the configured IP address of the RIST receiver and a user-selected UDP destination port P, where P is an even number between 2 and 65534. RIST receivers shall listen on UDP port P for media. This transmission is unidirectional, from sender to receiver. The sender may choose any arbitrary source port M for the RTP flow.
2. RIST senders shall periodically transmit the compound RTCP packets specified in section 5.2.1 to the configured IP address of the RIST receiver and UDP port P+1. The sender may choose any arbitrary source port R for the RTCP packets. RIST senders shall listen on port R for RTCP packets from the RIST receiver.
3. RIST receivers shall listen on UDP port P+1 for RTCP packets from the sender. These packets will have a source IP address S and a source port R' (in the absence of NAT devices between the sender and receiver, R and R' will be the same). RIST receivers shall send the RTCP packets they generate to IP address S and destination UDP port R', with a source UDP port of P+1. A receiver shall use S and R' from the last valid RTCP packet it has received from the sender.
4. RIST senders may offer the user the ability to manually configure source ports M and R.

These choices simplify the interaction of the RIST senders and receivers with NAT firewalls, as follows:

- If the RIST sender is behind a NAT device, the outgoing RTP and RTCP packets will establish state in the device, allowing the RTCP packets from the receiver to come back to the sender.
- If the RIST receiver is behind a NAT device, only ports P and P+1 need to be opened for operation.
- Receiving RIST devices may use UPnP to automatically configure firewalls.

5.1.2 Multicast Port Assignments

RIST can also be applied to multicast environments, such as private or isolated networks, or networks connected with multicast-capable tunnels. In a multicast environment, RIST will follow the standard UDP port assignments as per RFC 3550:

1. RIST senders shall transmit the RTP media packets to a user-selected UDP destination port P, where P is an even number between 2 and 65534. RIST receivers shall listen on UDP port P for media. This transmission is unidirectional, from sender to receiver. The sender may choose any arbitrary source port M for the RTP flow.
2. RIST senders shall periodically transmit the compound RTCP packets specified in section 5.2.1 to UDP port P+1, and the same multicast destination address as the media. The sender may choose any arbitrary source port R for the RTCP packets. RIST senders shall listen on port P+1 for RTCP packets from the RIST receiver.
3. RIST receivers shall listen on UDP port P+1 and the same multicast IP address as the media for RTCP packets from the sender. RIST receivers shall send the RTCP packets they

generate to the same multicast address and destination UDP port P+1. The receiver may choose any arbitrary UDP source port for its RTCP packets.

4. RIST senders may offer the user the ability to manually configure source ports M and R. RIST senders may use R=P+1 for simplicity.

5.2 RTCP Support

RIST senders and receivers shall implement a minimal subset of RTCP as described in this section. For senders, RTCP is used primarily to keep state on NAT devices along the path. For receivers, RTCP is used primarily to request lost packet retransmissions. The additional information included in the RTCP packet may be used sender and receiver devices to achieve better network performance.

5.2.1 Compound RTCP Packets

Multiple RTCP packets can be concatenated without any intervening separators and sent out in one UDP payload. RIST senders and receivers shall always transmit compound RTCP packets in order to comply with RFC 3550. The combinations shall be:

- RIST Senders:
  - Sender Report (SR) Packet (section 5.2.2) **OR** Empty Receiver Report (RR) Packet (section 5.2.3)
  - Source Description (SDES) Packet with a CNAME field (section 5.2.5)
- RIST Receivers
  - Receiver Report (RR) Packet (section 5.2.4) or Empty Receiver Report (RR) Packet (section 5.2.3)
  - Source Description (SDES) Packet with a CNAME field (section 5.2.5)
  - NACK Packet (if required – section 5.3.1)

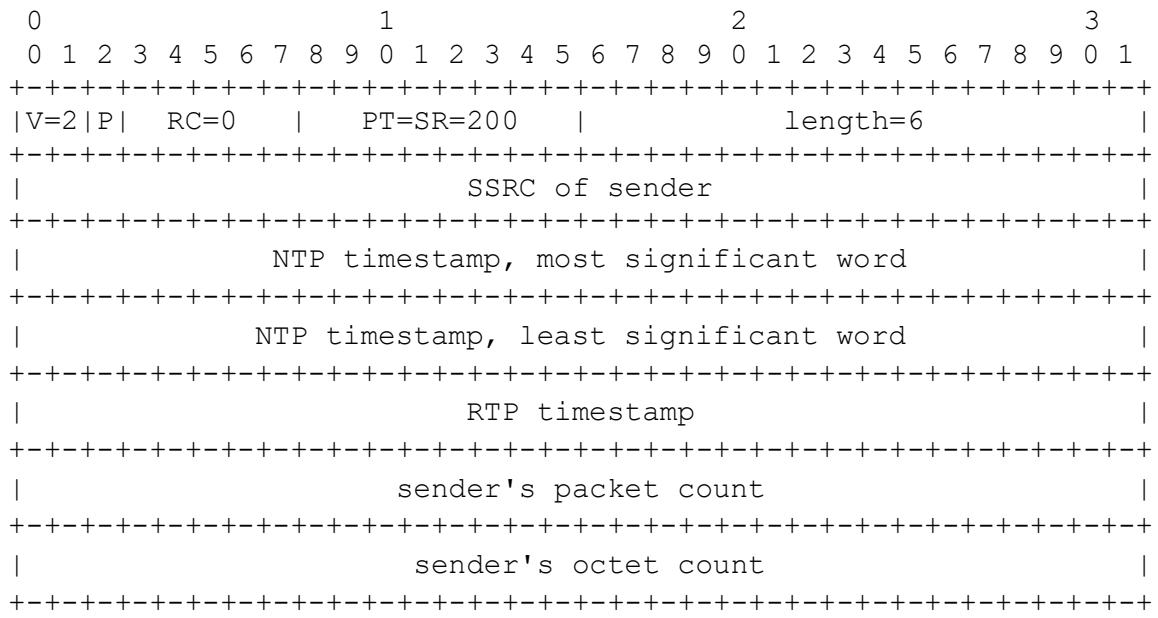
RIST sender and receiver devices shall transmit RTCP packets periodically. The rate at which the RTCP packets are sent must comply with the following requirements:

1. The interval between two successive RTCP packets shall be 100 milliseconds or less.
2. The maximum data rate used by RTCP packets shall be no higher than 5% of the average media data rate. For very low bit rate applications, requirement (1) above takes precedence.

The packet formats for all items are specified in the following sections.

5.2.2 Sender Report (SR) RTCP Packets

Since the primary use of RIST is in unicast links, the SR packet shall have no reception report blocks. The SR packet description presented below has been adapted from RFC 3550 section 6.4.1. RIST senders have the option of using the SR packet described in this section or an empty RR packet described later in this document.



The fields are used as follows:

version (V): 2 bits

Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RIST packets shall have V=2.

padding (P): 1 bit

Indicates whether or not there is padding at the end of the packet. RIST SR packets shall have P=0.

reception report count (RC): 5 bits

The number of reception report blocks contained in this packet. RIST SR packets shall have RC=0.

packet type (PT): 8 bits

Contains the constant 200 to identify this as an RTCP SR packet.

length: 16 bits

The length of this RTCP packet in 32-bit words minus one, including the header and any padding. RIST SR packets shall have length=6.

SSRC: 32 bits

The synchronization source identifier for the originator of this SR packet.

NTP timestamp: 64 bits

Indicates the wallclock time when this report was sent. The most significant 32 bits on this field indicate the number of seconds since 0h UTC on January 1900, and the least significant 32 bits indicate the fraction of the second. On a system

that has no notion of wallclock time but does have some system-specific clock such as "System uptime", a sender may use that clock as a reference to calculate relative NTP timestamps. A sender that has no notion of wallclock or elapsed time may set the NTP timestamp to zero.

RTP timestamp: 32 bits

Corresponds to the same time as the NTP timestamp (above), but in the same units and with the same random offset as the RTP timestamps in data packets. Note that in most cases this timestamp will not be equal to the RTP timestamp in any adjacent data packet. Rather, it shall be calculated from the corresponding NTP timestamp using the relationship between the RTP timestamp counter and real time as maintained by periodically checking the wallclock time at a sampling instant.

sender's packet count: 32 bits

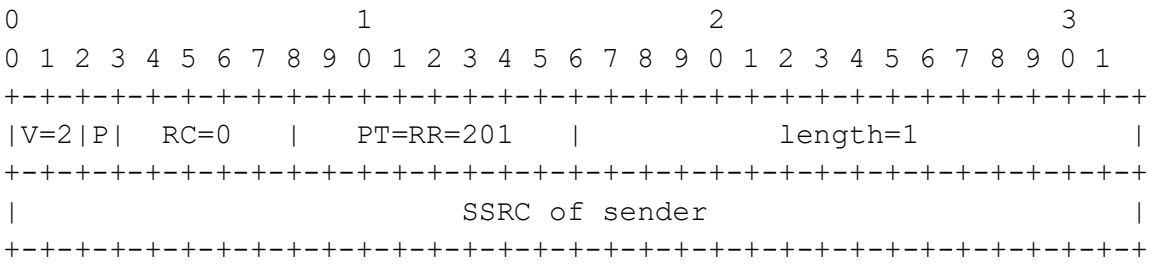
The total number of RTP data packets transmitted by the sender since starting transmission up until the time this SR packet was generated. The count should be reset if the sender changes its SSRC identifier.

sender's octet count: 32 bits

The total number of payload octets (i.e., not including header or padding) transmitted in RTP data packets by the sender since starting transmission up until the time this SR packet was generated. The count should be reset if the sender changes its SSRC identifier.

5.2.3 Empty Receiver Report (RR) RTCP Packets

Since the primary purpose of the RTCP transmission from the sender to the receiver is to possibly establish state in the firewalls along the path, the sender may elect to send an empty RR instead of the SR defined in the previous section. The empty RR is depicted below.



The fields are used as follows:

version (V): 2 bits

Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RIST packets shall have V=2.

padding (P): 1 bit

Indicates whether or not there is padding at the end of the packet. RIST RR packets shall have P=0.

reception report count (RC): 5 bits

The number of reception report blocks contained in this packet. Empty RR packets shall have RC=0.

packet type (PT): 8 bits

Contains the constant 201 to identify this as an RTCP RR packet.

length: 16 bits

The length of this RTCP packet in 32-bit words minus one, including the header and any padding. Empty RR packets shall have length=1.

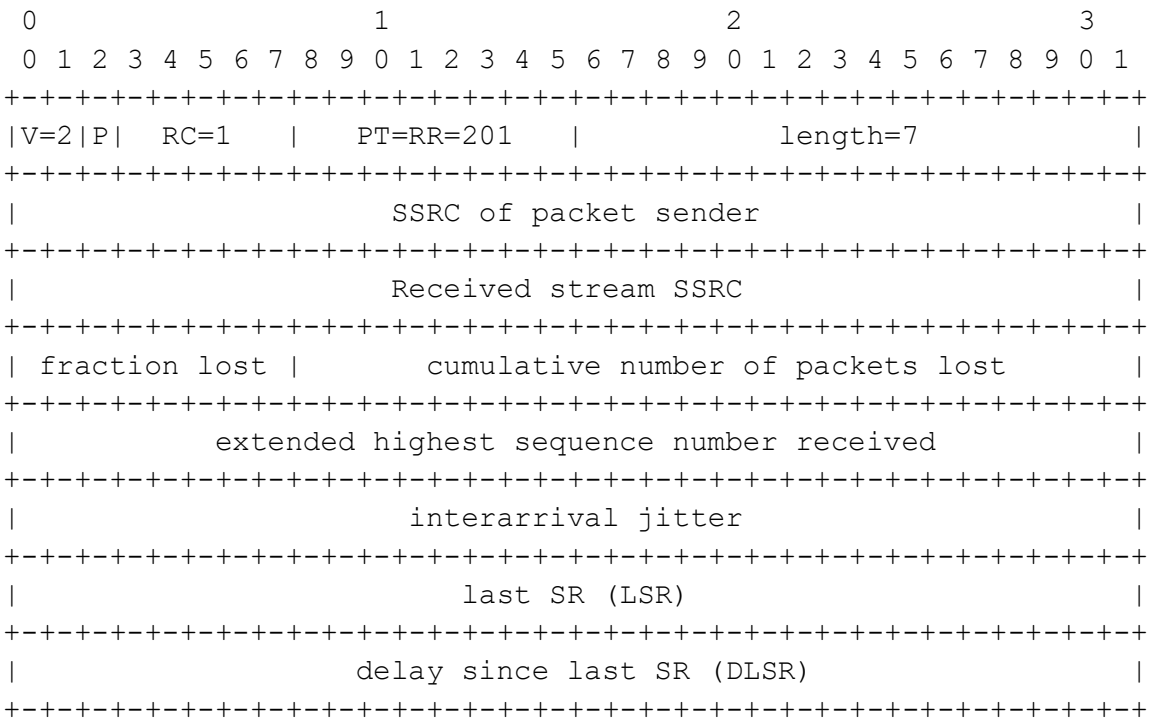
SSRC: 32 bits

The synchronization source identifier for the originator of this RR packet.

5.2.4 Receiver Report (RR) RTCP Packets

RIST RR packets shall have one and only one report block, corresponding to the RIST sender.

The SR packet description presented below has been adapted from RFC 3550 sections 6.4.1 and 6.4.2:



The fields are used as follows:

version (V): 2 bits

Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RIST packets shall have V=2.

padding (P): 1 bit

Indicates whether or not there is padding at the end of the packet. RIST RR packets shall have P=0.

reception report count (RC): 5 bits

The number of reception report blocks contained in this packet. RIST RR packets shall have RC=1.

packet type (PT): 8 bits

Contains the constant 201 to identify this as an RTCP RR packet.

length: 16 bits

The length of this RTCP packet in 32-bit words minus one, including the header and any padding. RIST RR packets shall have length=7.

SSRC of packet sender: 32 bits

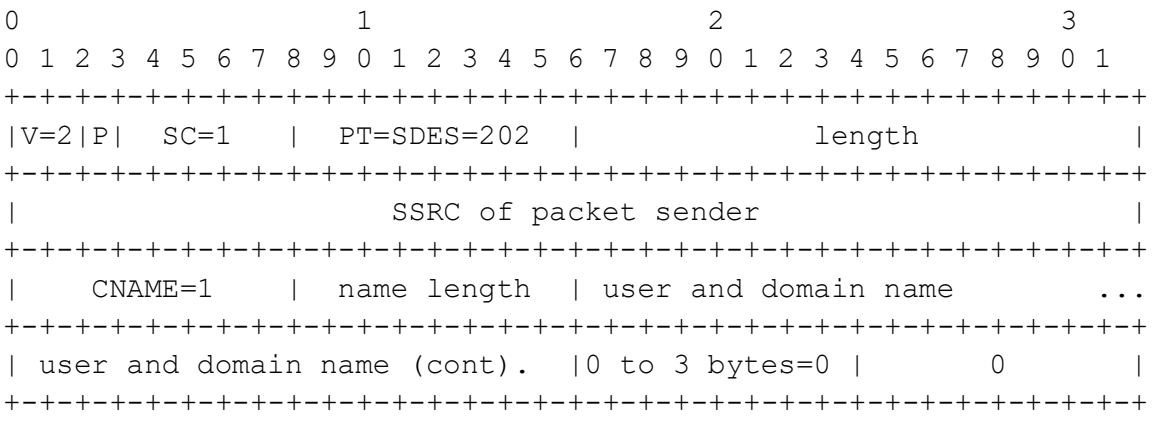
The synchronization source identifier for the originator of this RR packet.

fraction lost: 8 bits
cumulative number of packets lost: 24 bits
extended highest sequence number received: 32 bits
interarrival jitter: 32 bits
last SR timestamp (LSR): 32 bits
delay since last SR (DLSR): 32 bits

These fields shall be used as defined in RFC 3550 Section 6.4.1.

5.2.5 SDES RTCP Packets

RIST RTCP packets shall include one SDES packet with one item, the CNAME field. The complete SDES packet is shown below.



The fields are used as follows:

version (V): 2 bits

Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RIST packets shall have V=2.

padding (P): 1 bit

Indicates whether or not there is padding at the end of the packet. RIST SDES packets shall have P=0.

source count (SC): 5 bits

The number of chunks contained in this packet. RIST SDES packets shall have SC=1, corresponding to a CNAME field.

packet type (PT): 8 bits

Contains the constant 202 to identify this as an RTCP SDES packet.

length: 16 bits

The length of this RTCP packet in 32-bit words minus one, including the header and any padding.

SSRC of packet sender: 32 bits

The synchronization source identifier for the originator of this SDES packet (the RIST sender or receiver).

CNAME identifier: 8 bits

Identifies this item as a CNAME. Always set to 1.

name length: 8 bits

Length, in bytes of the user and domain name field.

The user and domain name is an ASCII string, which is **not** null-terminated as the length is specified in the previous field. RFC 3550 recommends that this string be programmatically generated in the form of "user@host". RIST implementations are free to use this field as they see fit. They may use the IP address of the sending or receiving device, in ASCII format (e.g., "192.168.129.10"). If the RIST device is multi-homed, it may use any of its local IP addresses. A RIST device shall make no assumptions about the data in this field, but a vendor may make use of this for proprietary purposes. If a vendor does so, they shall have a means of disabling this extension.

The SDES packet is terminated with a set of bytes set to zero, as follows:

- There needs to be at least one byte set to zero at the end of the SDES packet, not included in the CNAME name length.
- The overall size of the SDES packet needs to be a multiple of 32 bits.

Therefore, between 1 and 4 zero bytes will be added at the end of the SDES packet. These are not null terminators for the string - they indicate the end of the chunk list item in the SDES packet.

5.3 NACK-Based Recovery Protocol

RIST shall use a NACK-based Selective Retransmission protocol to recover from packet loss. The general operation of this protocol is as follows:

- Once packet loss is detected, receivers will request a retransmission of the lost packet or packets
- Receivers will implement a buffer to accommodate one or more network round-trip delays and packet re-ordering
- Packets may be requested multiple times

The receiver data flow is indicated below, and will be as follows:

- Packets will be received in a Reorder Section, which takes care of out-of-order packets. This will also support bonding of multiple channels (e.g., cell bonding).
- Packets will then cross into a Retransmission Reassembly Section. Packet losses will be detected at the boundary of these two sections by looking at discontinuities in the RTP sequence number.
- The buffer works as a FIFO, and an arriving in-order packet will push packets one or more positions in the FIFO. Out of order packets with sequence numbers between the newest packet in the reorder buffer and the oldest packet in the retransmission reassembly buffer shall be placed in positions according to their sequence number in the buffer.
- The decision of where in the buffer packet loss is detected is left up to the implementation. An implementation targeted at providing minimum possible delay will detect packet losses at the input of the buffer, and packets arriving out-of-order will cause extra retransmissions. Conversely, an implementation supporting bonding of multiple links will have a large enough reorder section to accommodate the worst case delay differential between the paths.

The protocol shall make provisions for networks where short signal outages may happen, and thus the buffer size will be a function of the round-trip time, packet jitter, and these expected outages. For the Simple Profile, the buffer size shall be manually configured at both sending and receiving ends.

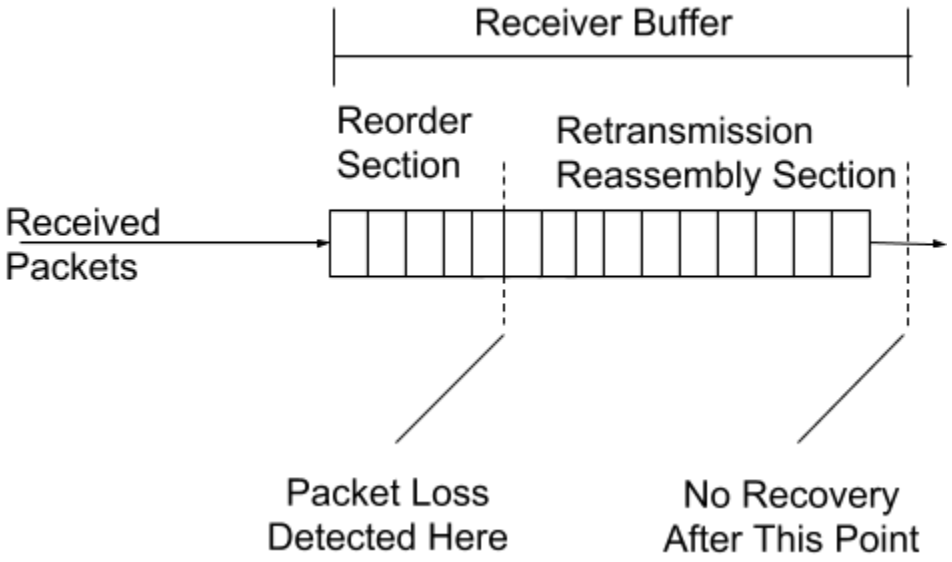


Figure 1: Receiver Buffers

5.3.1 Retransmission Requests

RIST Simple Profile includes two types of retransmission requests:

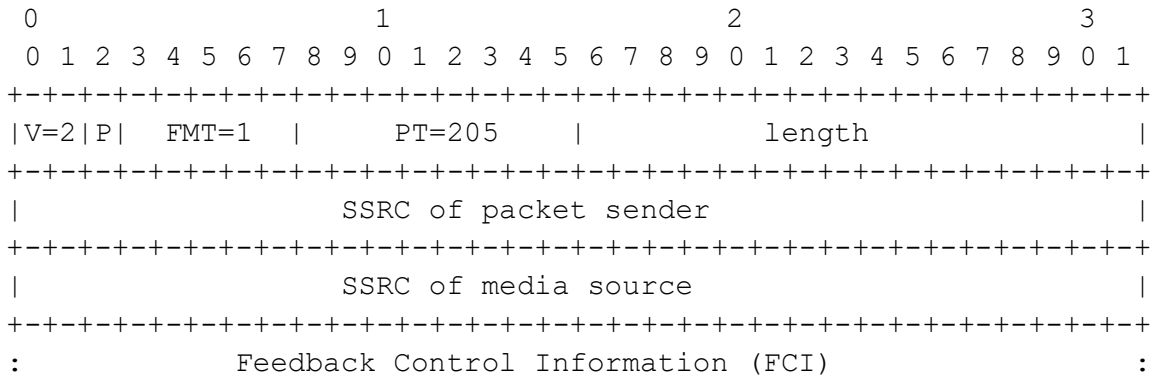
- A bitmask-based retransmission request, which is appropriate for individual packet losses and short loss bursts.

- A range-based retransmission request, which is appropriate for block losses.

RIST senders shall support both types of requests. RIST receivers may choose to implement either method, or both.

5.3.1.1 Bitmask-Based Retransmission Requests

Bitmask-based retransmissions shall be requested using the **Generic NACK** Message defined in RFC 4585 sections 6.2 and 6.2.1. The Generic NACK message specifies one or more ranges of 17 packets, using a bitmask to indicate which packets have been lost within each range. The message also contains the SSRC of the stream being requested. This is used to enable the sender to identify the flow in question. The retransmission request packet is formatted as follows:



The fields are used as follows:

version (V): 2 bits

Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RIST packets shall have V=2.

padding (P): 1 bit

Indicates whether or not there is padding at the end of the packet. RIST Generic NACK packets shall have P=0.

Feedback message type (FMT): 5 bits

This field identifies the type of the FB message and is interpreted relative to the type (transport layer, payload-specific, or application layer feedback). RIST messages shall use the Generic NACK code (1).

Payload type (PT): 8 bits

This is the RTCP packet type that identifies the packet as being an RTCP FB message. RIST messages shall use the code for Transport-Layer FB message (205).

Length: 16 bits

The length of this packet in 32-bit words minus one, including the header and any padding. This is in line with the definition of the length field used in RTCP sender and receiver reports. RIST messages shall use the value of **n+2**, where **n** is the number Generic NACK fields included in this packet, as specified below.

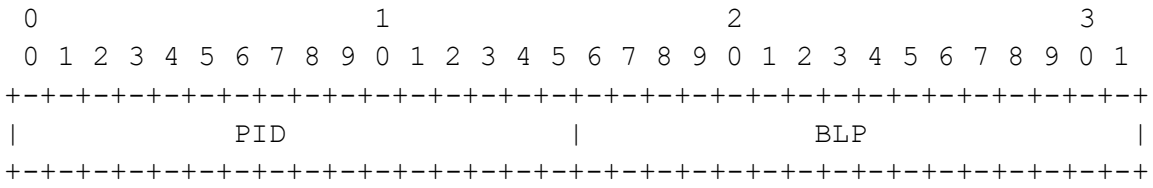
SSRC of packet sender: 32 bits

The synchronization source identifier for the originator of this packet. This field shall be ignored by the RIST sender.

SSRC of media source: 32 bits

The synchronization source identifier of the media source that this feedback request is related to. As indicated later in this document, the LSB of the SSRC is used to differentiate between original packets and retransmitted packets. The RIST receiver may use either value in the request packet.

Feedback Control Information (FCI): This field contains one or more instances of the 32-bit Generic NACK message shown below. Each FCI can request up to 17 lost packets. A Generic NACK message may contain multiple FCI fields.



The fields in the FCI are:

Packet ID (PID): 16 bits

The PID field is used to specify a lost packet. The PID field refers to the RTP sequence number of the lost packet.

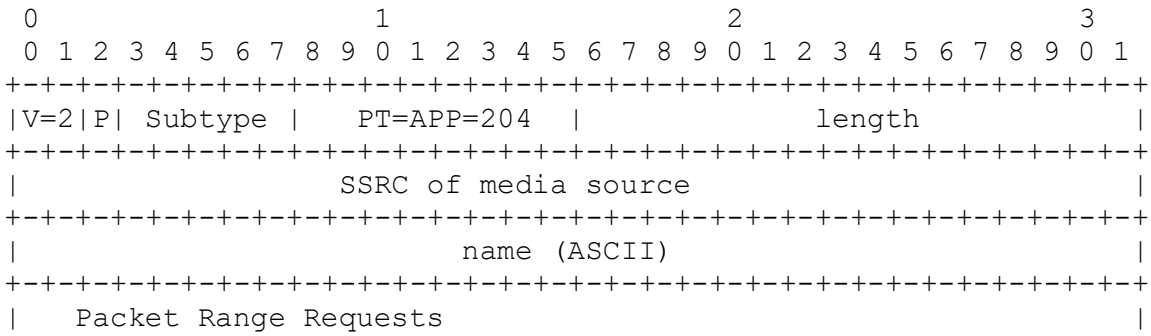
bitmask of following lost packets (BLP): 16 bits

The BLP allows for reporting losses of any of the 16 RTP packets immediately following the RTP packet indicated by the PID. Denoting the BLP's least significant bit as bit 1, and its most significant bit as bit 16, then bit i of the bit mask is set to 1 if the receiver has not received RTP packet number (PID+i) (modulo 2^16) and indicates this packet is lost; bit i is set to 0 otherwise. Note that the sender must not assume that a receiver has received a packet because its bit mask was set to 0. For example, the least significant bit of the BLP would be set to 1 if the packet corresponding to the PID and the following packet have been lost. However, the sender cannot infer that packets PID+2 through PID+16 have been received simply because bits 2 through 15 of the BLP are 0; all the sender knows is that the receiver has not reported them as lost at this time.

If a feedback message has multiple generic NACK fields, the fields should not overlap. In other words, any given sequence number, if included in the message, should fall into a single generic NACK field.

5.3.1.2 Range-Based Retransmission Requests

The range-based retransmission requests shall be implemented using RTCP. Ideally, this should be a Transport-Based Feedback Message (RTP type 205), with a new FMT value allocated by IANA. Such an allocation may be pursued by the RIST group in the future. In order to expedite the initial implementations of the protocol, the range-based retransmission request will use an Application-Defined RTCP message (RTP type 204), shown below.



The fields are used as follows:

version (V): 2 bits

Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RIST packets shall have V=2.

padding (P): 1 bit

Indicates whether or not there is padding at the end of the packet. RIST Generic NACK packets shall have P=0.

Subtype: 5 bits

This field identifies the type of the message. RIST range messages shall use code 0.

Payload type (PT): 8 bits

This is the RTCP packet type that identifies the packet as being an Application-defined Message, with PT=204..

Length: 16 bits

The length of this packet in 32-bit words minus one, including the header and any padding. This is in line with the definition of the length field used in RTCP sender and receiver reports. RIST messages shall use the value of **n+2**, where **n** is the number Range Request fields included in this packet, as specified below. A packet shall have a maximum of 16 range requests.

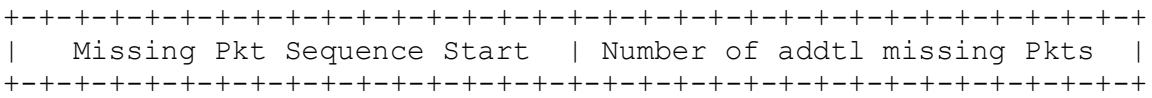
SSRC of media source: 32 bits

The synchronization source identifier of the media source that this feedback request is related to[1]. As indicated later in this document, the LSB of the SSRC is used to differentiate between original packets and retransmitted packets. The RIST receiver may use either value in the request packet.

Name (ASCII): 32 bits

This field identifies the application. For RIST packets, it shall have the value 0x52495354, the ASCII codes for "RIST".

Packet Range Requests: these are 32- bit fields, each requesting one packet range. The RTCP packet may contain multiple packet range requests. The packet range requests are shown below.



The fields in the Range Request are:

Missing Packet Sequence Start (16 bits):

RTP sequence number of the first packet dropped in the block

Number of Additional Missing Packets (16 bits):

Number consecutive packets being requested **after** the packet identified by the missing packet sequence start. For example, the Missing Packet Sequence Start is **N** and the Number of Additional Missing Packets is **A**, this indicates that packets from **N** to **N+A** inclusive have been lost. If **A** is zero, then only one packet (with sequence number **N**) is being requested.

5.3.1.3 RTCP Packet Size Considerations (Informative)

Both the Bitmask and Range RTCP NACK packets can contain multiple requests. It is a well-known fact that, when congestion occurs, smaller packets have a higher probability of being delivered. While this specification places no limits on how many requests can be included in a single RTCP NACK packet, it is recommended that implementers restrict the number of such requests. A limit of no more than 16 requests per packet is suggested.

5.3.2 Retransmitted Packets

For the retransmission of the requested data, the sender shall simply resend a copy of the requested packet, using the same transmission method as the other packets from this flow (i.e., the other packets with the same SSRC). The sender shall use the SSRC field to identify the flow

[1] RFC 3550 defines this field simply as SSRC/CSRC, and it was originally intended to contain the SSRC of the packet sender. However, this specification is re-defining this field as the SSRC of Media Source, since the SSRC of the packet sender is already available in the RR part of the compound RTCP packet, and is not useful in the RIST environment. Additionally, RFC 3550 indicates that APP packets with unknown names should be discarded, so non-RIST receivers will simply ignore this packet.

for which the retransmission is being requested. The retransmitted packet will have the exact same sequence number and timestamp, and will be identified by the least significant bit of the SSRC field, as follows:

- SSRC LSB=0: Original packet
- SSRC LSB=1: Retransmission packet

The remaining 31 bits of the SSRC will be the same between the original and retransmitted packets. In a multi-flow situation, this allows the receiver to match retransmissions to original flows.

The retransmission packet will be transmitted in the same method as the other packets in that same stream. If the flow is being sent to a specific IP address, then the retransmission packet will also be sent to that address; if the flow is using bonding technology (being split and transmitted to multiple IP addresses in parallel), then the same algorithm that is used to determine the next IP address will be used to determine where the retransmission packet will be sent. This will ensure that, whatever the path, the receiver will get the copy of the packet. If the stream is being transmitted to a multicast address, all the other receivers may be able to benefit from receiving a copy.

5.3.3 Burst Control (Informative)

Packet bursts may cause additional packet losses in a typical network. RIST implementations should manage packet bursts in the following two situations:

- NACK packet bursts (both Bitmask and Range modes): if a RIST receiver needs to send a large number of back-to-back NACK packets, care should be exercised in not creating too large of a burst. Depending on the loss pattern, one or the other modes may be more appropriate.
- Retransmitted packet bursts: a RIST sender may receive a NACK packet requesting a large number of retransmissions. It should exercise care in satisfying such a request as to not overload the network.
    - Note it is possible to generate a range-based retransmission request that requests the retransmission of every possible RTP sequence number (by setting the "Missing Pkt Sequence Start" field to any value, and the "Number of addtl missing Pkts" field to 65535).

For RIST Simple Profile, details of these techniques are left to the discretion of the implementers.

5.3.4 SSRC Filtering (Informative)

Both Bitmask and Range RIST packets contain the "SSRC of Media Source" field. Senders originating multiple streams may use this field to match requests with streams. A stream can usually be identified based on the IP addresses (source/destination) and UDP port of the RTCP

packet, except when a source is sending multiple streams (differentiated by SSRC) using the same address and port. The stream identification strategy is left at the discretion of the implementer.

5.4 Bonding Support

RIST Simple Profile supports bonding of multiple transmission channels (such as WiFi, LTE, etc.), as follows:

- An individual RTP media stream can be split between multiple channels in order to combine their bandwidths.
- An individual RTP media stream can be replicated between multiple channels in order to increase reliability.

Both techniques can be used simultaneously. In these cases, receivers will need to combine the packets in order to reconstruct the original media stream.

RIST supports bonding as follows:

- A RIST receiver may listen to multiple RTP and corresponding RTCP packets over one or more channels. it is the responsibility of the RIST receiver to re-aggregate the packets to one buffer.
- The user may configure the RIST receiver listen to multiple RTP and RTCP sessions in parallel.
- The RIST Receiver should send RTCP reports to all sessions associated with one stream.
- The RIST Receiver may select send NACK reports to one or more RTCP sessions
- The RIST receiver may decide to spit / duplicate / prioritize the NACK reports based on the session.
- The RIST sender shall listen to the NACK reports and send retransmission packets according to the session and source
- The RIST sender may choose to block/deny a session from packet retransmission (for demo purposes, or preserve bandwidth).
- The RIST RTP sender may duplicate/split the RTP stream among the available sessions.

A session may be multicast, unicast and any combination as long as the session can support the chosen method.

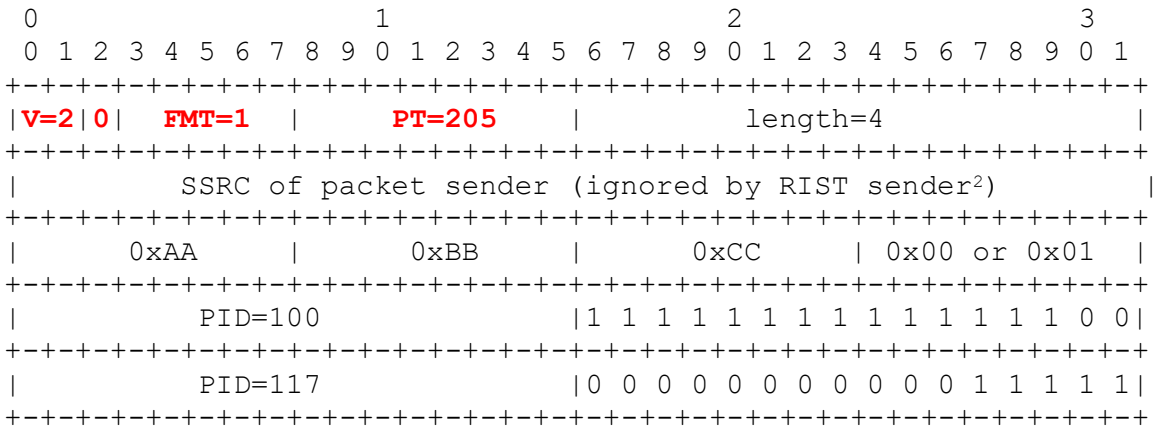
A SMPTE-2022-7 Class-C compliant receiver may be able to receive a RIST bonding stream (without the packet retransmission capabilities) if the Path Differential falls within the limits indicated for that class.

Appendix A (Informative) RIST Retransmission Request Examples

This section is informative. Its purpose is to provide NACK packet examples to aid in the preliminary implementations. In the examples below, numbers prefixed with 0x are hexadecimal, and numbers without that prefix are decimal.

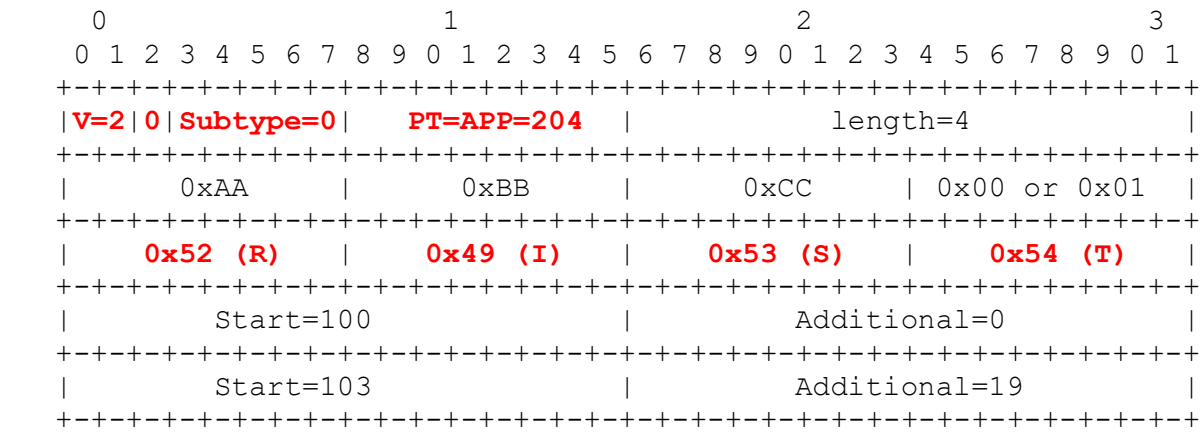
For this example, assume that the RIST device is receiving a stream from IP address 192.168.1.10, UDP port 3000, SSRC 0xAABBCC00. The RIST device correctly receives packet sequence number 99, then misses packet 100, then receives packets 101 and 102, then misses the following 20 packets, from 103 to 122, and then receives all subsequent packets starting at sequence number 123.

The Bitmask-Based retransmission request packet will be sent to 192.168.1.10 port 3001 with the following contents:



The Range-Based retransmission request packet will be sent to 192.168.1.10 port 3001 with the following contents:

[2] RFC 3550 requires senders to keep an SSRC list, so an implementation that is fully compliant with that RFC will need to process this field. However, it is not required for RIST operation, so a RIST sender may choose to simply ignore this field.



In both examples above, the contents of the fields in **red** are fixed by this standard and never change.

Appendix B (Informative) Suggested Default Values

RIST implementations complying with this specification are manually configured by the user. In the absence of user input, the following default parameters are suggested:

- Receiver Buffer: 1000 milliseconds
- Sender Buffer: equal or higher than receiver buffer
- Reorder Section: 70 milliseconds
- Number of Retransmission Requests per Packet: 7

The interval between retransmission requests can be derived from these parameters. It is the receiver buffer minus the reorder section divided by the number of retransmission requests. For the above values, the outcome is 132 milliseconds.