Video Services Forum (VSF)

Reliable Internet Stream Transport (RIST) ARQ Best Practices

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Executive Summary

Packet recovery for the RIST Series of Technical Recommendations uses on-demand retransmission of dropped packets, a technique known as ARQ. RIST implementations may execute the protocol functions in any compliant way. This document describes the best practices when implementing ARQ to achieve good packet recovery without congesting the network. The recommendations included here are applicable to any ARQ-based system, not just RIST.

Recipients of this document are invited to submit technical comments. The VSF also requests that recipients notify us 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 best practices described in this document, and to provide supporting documentation.

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1 Introduction

As broadcasters and others 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 protocol solutions are currently available on the market; however, incompatibility exists 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 using existing or new standards and recommendations.

RIST packet recovery uses the well-known ARQ method (also known as Selective Retransmission): the receiver only requests the packets that are dropped or delayed in transit. The RIST Technical Recommendations only describe the protocol. Implementers are free to make decisions on buffer sizes, when to request packet retransmissions, and how often to send replacement packets. This document describes a set of best practices when implementing a RIST system. These best practices recommend buffer sizes and other parameter values that provide a good balance between packet recovery, network congestion control and added delay.

The recommendations presented in this document are applicable to any ARQ-based packet recovery system. They are based on decades of collective experience with real-world implementations of ARQ across a wide variety of network distances and performance levels.

In some cases, the recommendations in this document refer to more advanced RIST features that can be combined with ARQ to provide further improvements. Other ARQ-based protocols may or may not have these features.

1.1 Contributors

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

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1.2 About the Video Services Forum

The Video Services Forum, Inc. (www.vsf.tv) 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 e-mail opsmgr@vsf.tv.

2 Conformance Notation

This document is informative and there are no normative sections.

3 References

VSF TR-06-1:2020, Reliable Internet Stream Transport (RIST) Protocol Specification – Simple Profile

VSF TR-06-2:2024, Reliable Internet Stream Transport (RIST) Protocol Specification – Main Profile

VSF TR-06-3:2024, Reliable Internet Stream Transport (RIST) Protocol Specification – Advanced Profile

VSF TR-06-4 Part 1:2022, Reliable Internet Stream Transport (RIST) – Source Adaptation

VSF TR-06-4 Part 6:2024, Reliable Internet Stream Transport (RIST) – Transport Stream Program Selection

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

Any mention of references throughout the rest of this document refers to the versions described here, unless explicitly stated otherwise.

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4 ARQ Best Practices

4.1 Best Practices for Buffer Configuration

Packet buffers in both Sender and Receiver devices are required for any ARQ system. Choosing an appropriate receiver buffer size is the key to optimizing ARQ system performance for two reasons:

1) If the buffer is too small, the system will be less able to correct errors that could be fixed and thus be less useful on poor-quality (lossy) links or less able to withstand a transient link error state.
2) If the buffer is too large, the system will have excessive end-to-end latency, making the system undesirable for time-critical applications,

Two buffers are typically used in an ARQ transmission system, one located in the Sender and one in the Receiver (which is further divided into two subsections). These buffers are:

1) The Sender Retry Buffer is used by the Sender to store packets that might be needed to fulfill a request for retransmission. Note that this buffer DOES NOT impact the end-to-end latency of the ARQ link and can be made as large as desired. However, this buffer should not be smaller than the product of the Round-Trip Time (RTT) multiplied by the maximum number of ARQ retries of any receiver plus one.
2) The Receiver Buffer is used by the Receiver to create the best possible set of correctly sequenced transport stream packets (or other data packets in Advanced Profile) that can then be consumed by other Receiver processes, such as an MPEG decoder. This buffer is conceptually divided into two sections: the Reorder Section and the Retransmission Reassembly Section:
    a. The Reorder Section (also called a Packet Reorder Buffer) is used in the receiver to decide whether a given packet needs to be requested (or re-requested) from the sender. This buffer should be large enough to absorb any packet jitter (also called packet delay variation) that is present on the link as well as accommodating any packet reordering that has occurred during transmission from the sender to the receiver. In general, a receiver should NOT send a packet retransmission request until after the Reorder Section buffer time has expired for any missing incoming packet, whether it is an original packet or one that has been retransmitted.
    b. The Retransmission Reassembly Section is used to put received packets/data into the correct sequence before consumed by other receiver processes. Correct bitstream order is crucial for MPEG decoders and other processes that are not made to handle bitstreams that are jumbled. ARQ processes need a way to insert retransmitted packets into the correct sequence with packets that are received correctly on the first try, so this section of the receiver buffer provides storage necessary for the reassembly to take place.

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In other words, the Reorder Section can be seen as a "grace period" before a retransmission is requested. For example, if packet with sequence number 6 is received after sequence number 4, the Receiver does not immediately request packet sequence number 5; it waits for the reorder time to give packet 5 time to arrive. In the case of RIST, this is described in Section 5.3.1 of TR-06-1.

Table 1 lists recommended parameter settings for applications operating over high-quality links that are sensitive to delay, such as live sports and interactive news programming. Table 2 lists recommended parameter settings for applications operating over average-quality networks that are less sensitive to delay, such as bonded cellular, cloud access and streaming applications. Buffer sizes are typically a multiple of the expected RTT, which can be measured on the link or calculated. In the case of RIST, the RTT Echo Request/Response packets from TR-06-1 section 5.26 can be used to measure the actual RTT, using the same packet size as the data. The values provided in these tables are intended to provide a useful starting point; system operators should configure their ARQ parameters based on their own RTT measurements, Note that once a buffer size is configured it generally remains fixed for the duration of any RIST session; changing the size of the buffer during use could cause downstream processes to lose lock or experience timing problems.

Table 1: Low-Latency (Live) Operation Parameters

Profile	**Typical RTT**	**RTT Lower Boundary Setting**	**RTT Upper Boundary Setting**	**Reorder Section Buffer Size**	**Retransmission Reassembly Buffer Size**	**Notes**
Short Haul < 500 km	11-25 ms	12 ms	25 ms	2-3 ms	25-50 ms	1-2 retries/packet, good quality link
Medium Haul 500-3000 km	50-100 ms	20 ms	100 ms	3-7 ms	50-150 ms	1-2 retries/packet, good quality link
Long Haul >3000 km	250-350 ms	250 ms	400 ms	5-10 ms	400-1200 ms	2-3 retries/packet, good quality link

Table 2: Assured Delivery Operation Parameters

Profile	**Typical RTT**	**RTT Lower Boundary Setting**	**RTT Upper Boundary Setting**	**Reorder Section Buffer Size**	**Retransmission Reassembly Buffer Size**	**Notes**
Short Haul < 500 km	11-25 ms	12 ms	25 ms	2-5 ms	50-125 ms	3-5 retries/packet, standard Internet
Medium Haul 500-3000 km	50-100 ms	20 ms	100 ms	3-10 ms	400-750 ms	5-7 retries/packet, standard Internet
Long Haul >3000 km	250-350 ms	250 ms	400 ms	5-20 ms	1000-4000 ms	7-10 retries/packet, standard Internet

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4.2 Best Practices for Both Senders and Receivers

- Include RTT boundaries, independent of the RTT measurement (pre-configure minimum and maximum RTT). These boundaries prevent a system from excessive retries in a low delay link and prevent excessively large buffers in a high delay link
- Ensure that sender and receiver have matching configuration parameters. The most important parameter is the buffer size. The sender buffer size must not be lower than the receiver buffer size. In a one-to-many situation, the sender buffer size needs to be larger than the largest receiver buffer size.
- Support for NULL Packet Deletion (NPD) is highly recommended as it reduces traffic, which in turn lowers the pressure on the network. In RIST, NPD is defined in TR-06-2 Section 8.
- Support for multi-path communication can greatly improve the reliability of an ARQ link:
    - The use of redundant links (ST 2022-7 seamless switching) is crucial to achieving high reliability.
        - Use distinct paths/ISPs for each of the links.
        - Configure the Reorder Section to be at least equivalent to the latency differential between the paths.
        - Only retransmit packets that are lost on all the links.
    - If using bonding (spreading traffic over multiple links), reliability can be improved by dynamic load sharing. For RIST, this is defined in TR-06-4 Part 1.
- If supported, drop lower-priority services if the available network bandwidth is exceeded, thereby reducing the stream bandwidth and alleviating congestion. These services can later be restored if the available network bandwidth improves. When using RIST, determination of which services to drop can be based on the link quality feedback messages from TR-06-4 Part 1 or based on explicit requests from the receiver using the Transport Stream Program Selection defined in TR-06-4 Part 6.

4.3 Best Practices for Senders

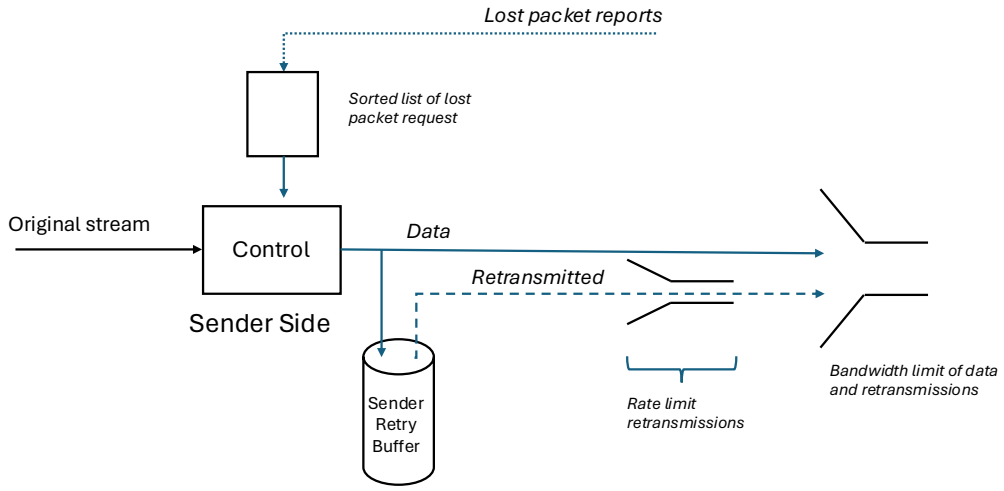
- Implement an optional retransmission bandwidth limit, which should be less than or equal to the amount of available end-to-end bandwidth on the network minus the amount of raw traffic. Pace the retransmission packets to not exceed this limit, avoiding sender bursts (which can cause downstream packet drops due to limited router buffer sizes). This is illustrated in Figure 1.
- Space each retransmission of a given packet by one RTT plus a small delta (in case the receiver is too aggressive).

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- Use a sorted list by request sequence numbers instead of a simple FIFO for the retries. This sorts the retransmission requests by oldest sequence number first; newer requests receive lower priority.
- If there is a retransmission bandwidth limit and retransmissions need to be dropped, drop the newer requests first.
- If two similar sequence number requests are received very close to each other (less than half an RTT) from the same receiver, drop one of them, as the receiver is being too aggressive.
- If the sender is an encoder capable of bit rate changes on-the-fly, change the bit rate based on network conditions. For RIST, this is defined in TR-06-4 Part 1 Source Adaption.

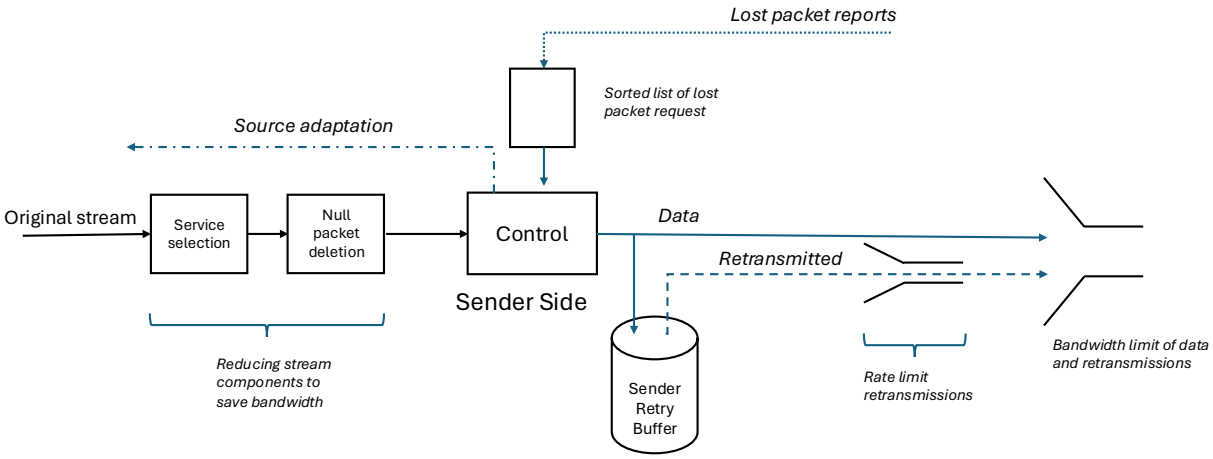
The sender advanced features (NPD, Source Adaptation) are illustrated in Figure 2.

Figure 1: Sender Side (basic operation)



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Figure 2: Sender Side (advanced features)



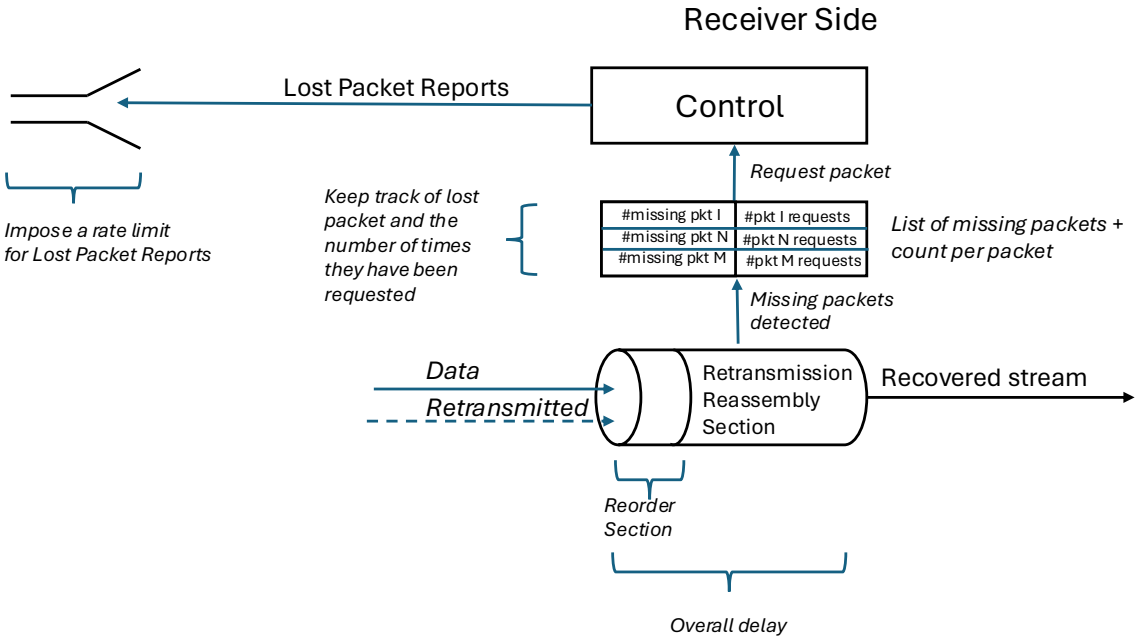
4.4 Best Practices for Receivers

- The receive buffer should be set to a multiple of the RTT value.
- Implement an optional cap on the bandwidth for retransmission requests, based on the return channel capacity. If the bandwidth cap is reached, wait for another RTT to send the request.
- Space retransmission requests for the same packet by one RTT plus a small delta. Retrying more often than this causes additional unnecessary retransmissions.
- Implement an optional maximum count for retransmission requests. This becomes useful when the network has a low RTT and the buffer is large. A recommended value for this maximum is on the order of 10 requests.
- Once a loss is detected, wait for half an RTT before requesting the packet (or your reorder window). This gives the packet additional time to arrive (it may just be late).
- Maintain an ordered list of lost packets requested and the associated time they were requested; remove list entries that have arrived or timed out.
- Do not request a retransmission of a missing packet when there is less than one RTT before the data on that packet is scheduled to be consumed. In this case, even if the retransmission arrives, it will arrive too late.
- The reorder buffer needs to be sized properly to prevent unnecessary retries. A symptom of a low reorder buffer is duplicate packets being received. A measurement of the inter-packet arrival time (IAT) can be used to size this buffer; the maximum IAT represents the lower boundary of the reorder buffer.

The receiver recommendations are illustrated in Figure 3.

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Figure 3: Receiver Side



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