RIST – Reliable Internet Stream Transport

RELIABLE INTERNET STREAM TRANSPORT

What is RIST?

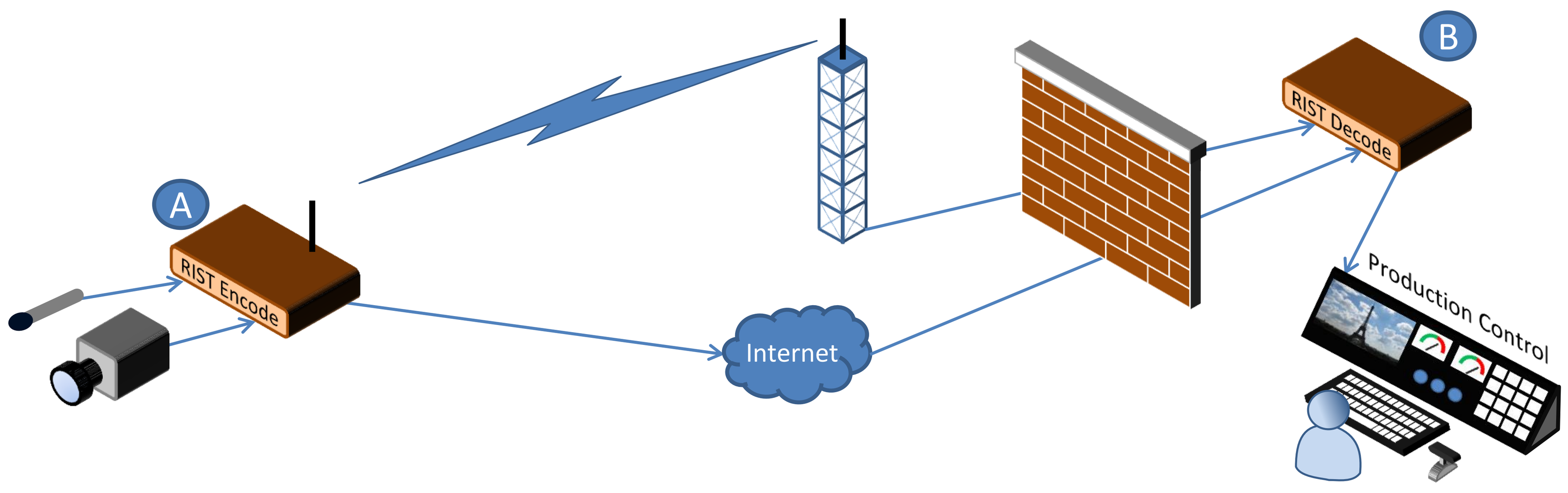
R.I.S.T. is a Video Services Forum effort to define and promote an interoperable standard for the transport of live video content in real time at low latency over unmanaged networks, including the Public Internet. Currently there are many successful solutions available to achieve this goal but they are manufacturer-specific and they do not interoperate. A method is needed for unrelated manufacturers' solutions to interoperate and advancing this technology towards its full potential.

The Preliminary Goals:

Contribution Quality over Public Internet
Interoperable
Low Latency

Operates Over:

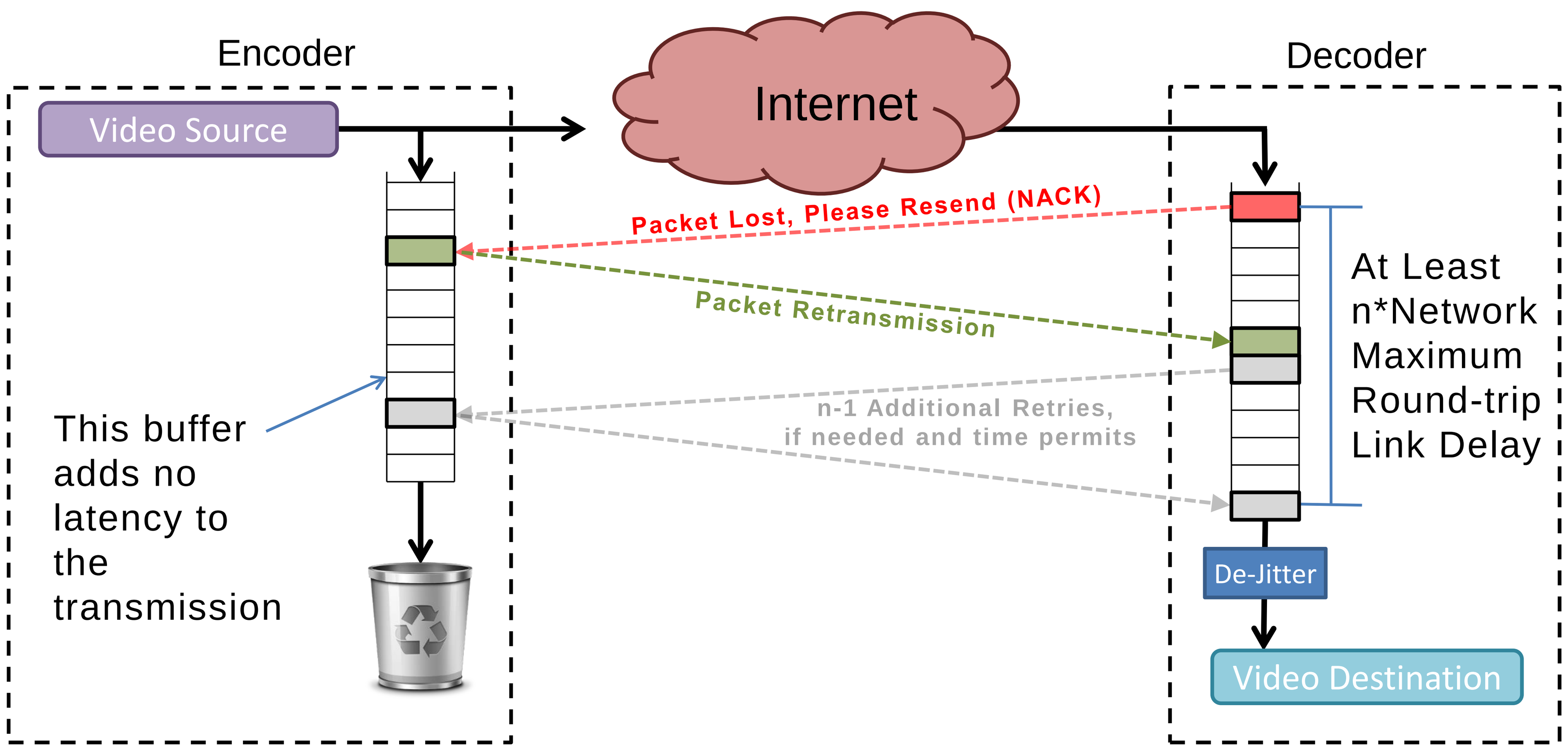
Hard-wired Internet
LTE / Bonded LTE
Satellite



A Basic RIST System

A basic RIST system consists of a Sender (A) and a Receiver (B) connected across a potentially lossy network, possibly with multiple transmission paths enabled.
RIST shall use a NACK-based Selective Retransmission protocol to recover from packet loss. The general operation of this protocol is as follows:

- Receivers do not communicate with senders unless they detect packet loss
- 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, provided that the decoder buffer is large enough to permit the recovered packet to be placed in the proper sequence in the decoder stream.



RIST Data Flow

The RIST Encoder is responsible for accepting a video input (basic version accepts RTP streams in multiple media formats) and generating an appropriately labeled output stream. No major modifications are made to the stream other than the RTP SSRC, destination IP address(es) and the UDP port number. Note that there should be nothing in the RIST that should prevent the use of multiple simultaneous IP connections.

As packets are transmitted, the Encoder stores a copy of each outbound packet for a period of time in case the Decoder requests a re-transmission of that packet. Note that this buffer does not add any latency to the overall end-to-end packet transmission time.

Packets can travel from the RIST Encoder to the RIST Decoder over standard unicast or multicast IP networks, which can be expected to occasionally experience packet loss.

The RIST Decoder is responsible for most of the processing and intelligence of the overall transmission system. As packets arrive at the Decoder, they will be received in a buffer that processes out-of-order packets and puts them back into the correct order according to their sequence numbers. This operation will also support bonding of multiple channels (e.g., cell bonding), allowing for the likely scenario that packets traveling along different paths will experience different amounts of delay and therefor arrive at different times. The length of this buffer will need to be at least as long as the difference between the best-case and worst-case path delays between the Encoder and the Decoder, with enough added margin to accommodate any packet reordering caused by the network.

The next major processing step is to analyze the RTP packet numbers and determine if any packets are missing from the stream by looking for gaps in the number sequence. If so, the Decoder will then need to make a request to the Encoder to have the missing packets retransmitted.

Packet Retransmission is initiated by the Decoder sending an (unsolicited) NACK RTCP packet to the Encoder. Data in this NACK (negative acknowledgement) packet indicates the sequence numbers of the missing packets. When this message is received by the Encoder, it retrieves the indicated packets from its buffer and re-sends them to the Decoder.

When the packets arrive at the Decoder, they must be put back into the proper sequence within the decoder buffer. This step is necessary in order for the output stream to properly reproduce the stream at its output. For this to be able to work, the buffer in the Decoder must be able to store packets for at least as much time as the round trip delay between the Encoder and the Decoder. (A full round trip is required for the RTCP NACK packet to travel from the Decoder to the Encoder, and then for the retransmitted packets to travel from the Encoder to the Decoder.)

Multiple round trips can be used in applications that are not particularly sensitive to delay. For these cases, the Decoder can send NACK messages to the Encoder multiple times to request missing packets. The number of re-tries (n) that can be made is limited by the size of the decoder buffer, which must be at least n times the maximum round-trip delay. Note that every packet must pass through the decoder buffer in FIFO fashion, which in turns dictates that the overall delay of the RIST system is driven by the duration of the decoder buffer. A de-jitter buffer is used at the Decoder output to smooth the flow of outbound packets.

Other Industry Standards and Related Work

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

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

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/AVPF)

IETF RFC 4588, RTP Retransmission Payload Format

The Impact of Lost Packets

Video streams, particularly compressed ones, can be severely disrupted by lost data packets. Visual and audio signal impairments, signal interruptions and even the complete disruption of a stream can be caused by the loss of a single packet. The table below shows how often a video signal would experience a disruption on a link without packet re-transmission technology at different packet loss ratios. The calculations in the table below were made using a 4 Mbps second stream with 1316 bytes of video payload in each packet.

Packet Loss Ratio	Drop one packet in	Produces a glitch every
10^{-3}	1,000	2.6 seconds
10^{-4}	10,000	26 seconds
10^{-5}	100,000	4 minutes 23 seconds
10^{-6}	1,000,000	44 minutes
10^{-7}	10,000,000	7 hours 19 minutes

RIST Team Points of Consensus

Base Protocol: RTP

- Protocol will follow whatever RTP profile is already in place for the type of media being carried
- RTP Payload Type must be handled properly by sender and receiver
- Timestamp built into the RTP header deemed sufficient for de-jitter, if required

Packet Recovery Method: selective retransmission (ARQ)

- Receiver will only send NACKs for packets it did not receive
- A lost packet may be requested multiple times within a user-defined interval
- Receiver will include provisions for packet re-ordering and de-duplication prior to requesting retransmission (to support bonded links and other multipath)

NACK requests will use RTCP messages

- Ongoing discussion to adopt Generic NACK message from RFC 4585
- Ongoing discussion for private metadata insertion

Retransmitted packets will be identified by a different SSRC (as per RFC 4588)

- Ongoing discussion of signaling to communicate this SSRC (Synchronization Source)

Current Discussion Items

Will RIST include VPN capabilities?

- Selection of VPN technology is under discussion

RIST will include encryption capabilities

- Selection of encryption technology is under discussion
- Encryption session establishment
- Key management (default key?)

Upcoming Discussion Items

Protocol auto-configuration:

- Connection/session initiation
- Auto-configuration of latency
- Auto-configuration of number of retransmissions
- Parameter negotiation

Media-specific optimizations (i.e., NULL packet removal)

Option to include FEC in the protocol

Application Layer interoperability

- Video/Audio Codec selection
- File/stream container formats
- Bit rates
- File/stream container formats

Interoperability Profiles

- Craw/Walk/Jog/Run

	User Requirements	Sender Features	Receiver Features
Crawl	Single packet loss recovery Short burst loss recovery	Fixed bit rate coding User-controlled settings	Fixed buffer size User-controlled settings
Walk	Long burst packet loss recovery	Stream negotiation In-band signaling	Adjustable buffer
Jog	Variable network bandwidth	Adjustable bitrate coding Network bandwidth probe Multipoint distribution	Bandwidth estimation Adaptive buffer
Run	Network link aggregation Redundant transmission paths	Multiple unicast streams IGMP multicasting Scalable coding	Hitless protection switch Scalable decoder

For More Information, Contact:

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http://vsf.tv/RIST.shtml

Actively Participating Companies