RIST Source adaptation

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What is the story with adaptivity?



- Most delivery protocols assume that the link capacity has enough capacity to support the stream and some extra head room for ARQ
- Most network do have their quirkiness events:
 - Over subscription
 - Sudden high interference
 - Capacity drop
 - Link saturation
- HTTP Adaptive streaming showed the world that this problem maybe overcome by using different profiles (different bitrates and resolutions) to provide a sustained delivery.
- How do we bring the same concept to Low delay transmission and take these challenges head-on? While providing the lowest delay possible.



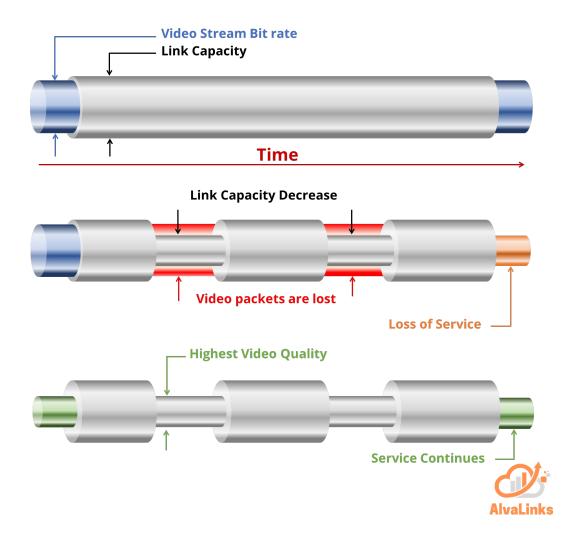
Reliable Video Delivery building blocks **P**SHOWCASE Adaptivity **RIST with source adaptation** Load Share Path Diversity **S** ~ **B** Security





Why do we need Source adaptation?

- Stream bitrate is known ahead of any streaming
- The link capacity should be slightly higher than the video stream's bit rate to allow packet loss protection
- The link capacity can change over time or fluctuate due to over subscription, interference, equipment failure
- The challenge is to find adapt the source stream to the available capacity stream to fit with the new link capability
- A receiver sends gathers statistics on the link; #packet loss, #incoming packets, #retransmitted packets and more to the Sender.
- The Sender can use the receiver reports to asses the link's capacity in real time and take action:
 - Adjust the rate with additional links
 - **Command the source to change the stream bit rate**
 - Change stream components



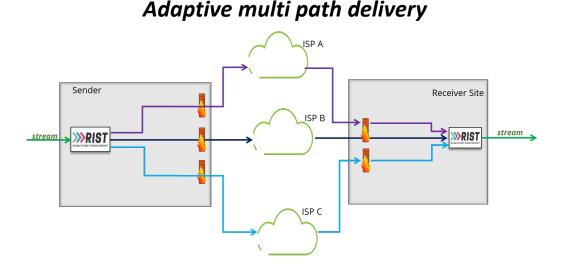
IOWCASE

Source adaptation support

Specification: TR-06-04 part 1 it covers this modifications:

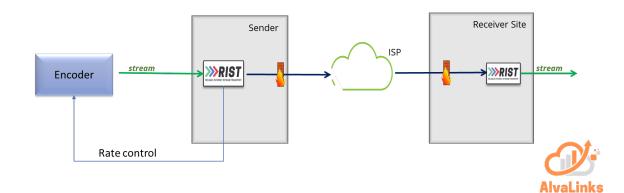
- TR-06-01 Simple profile through RTCP messages
- TR-06-02 carriage of TR-06-01 messages
- TR-06-03 Specific control messages

Today we are going to highlight two use cases:

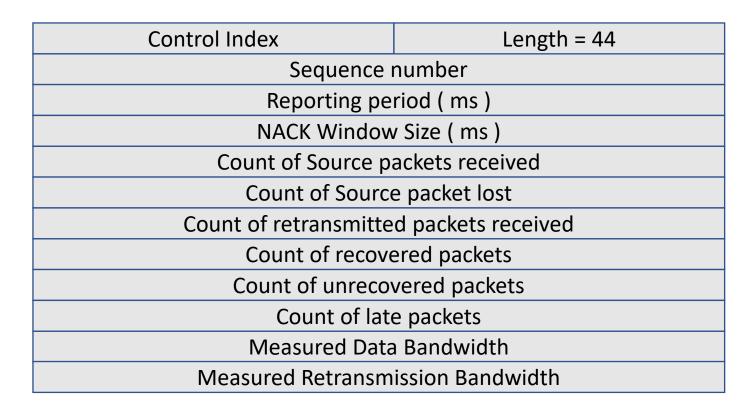


Adaptive source control

WCASE



RIST new Receiver report message





HOWCASE

Message components



Sequence Number	Increasing message sequence number. It allows the sender to detect lost or duplicate Link Quality messages
Reporting Period	Duration of the reporting period, in milliseconds
NACK Window Size	Current Receiver NACK window setting in the receiver, in milliseconds.
Count of source packets received	Number of source packets received during the reporting period
Count of original packets lost	Number of original packets lost during the reporting period
Count of retransmitted packets received	Number of retransmitted packets received during the reporting period
Count of recovered packets	Number of packets originally lost and then recovered through retransmission during the reporting period
Count of unrecovered packets	A count of unrecovered packets during the reporting period. These would be packets that were not recovered during the NACK window
Count of late packets	Count of source packets received too late to be used (i.e., outside the NACK window). More specifically, packets whose sequence number is earlier than the last packet released from the NACK buffer
Measured Data Bandwidth	This is the number of source payload data bits plus the RTP header bits (including any extensions) received during the reporting period in seconds and rounded to the closest 1000 bits/sec. Null packets are counted as full TS packets after NPD inflation. Measured in Kbits/sec
Measured Retransmission Bandwidth	This is the total number of retransmitted payload data bits plus the RTP header bits (including any extensions) received during the reporting period, divided by the reporting period in seconds and rounded to the closest 1000 bits/sec. Null packets are counted as full TS packets after NPD inflation. Measured in Kbits/sec

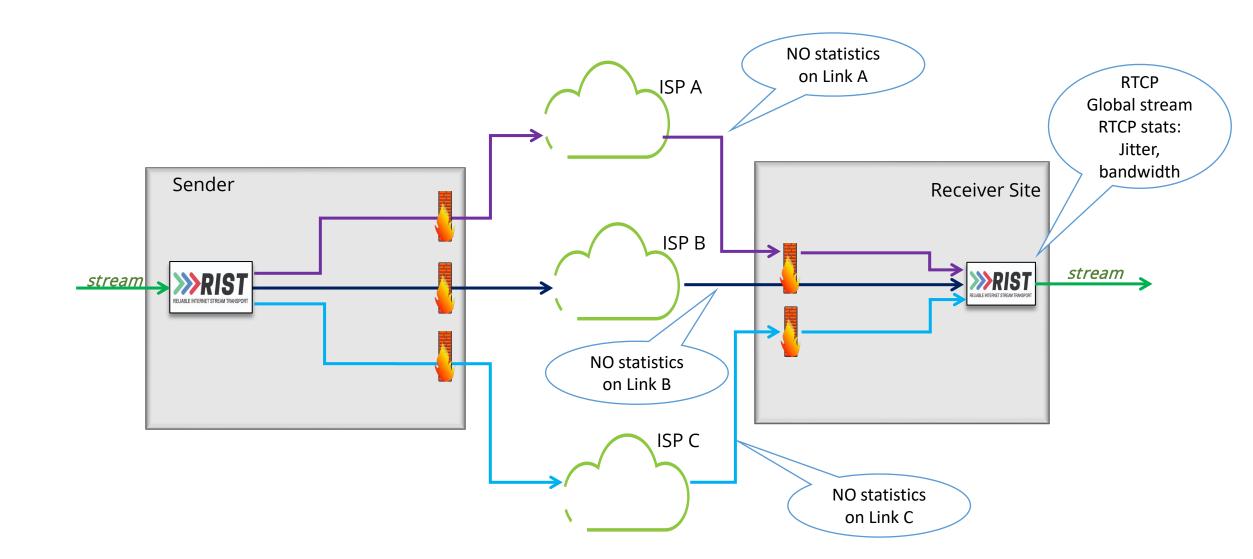
Why do we need this capability?

IP SHOWCASE

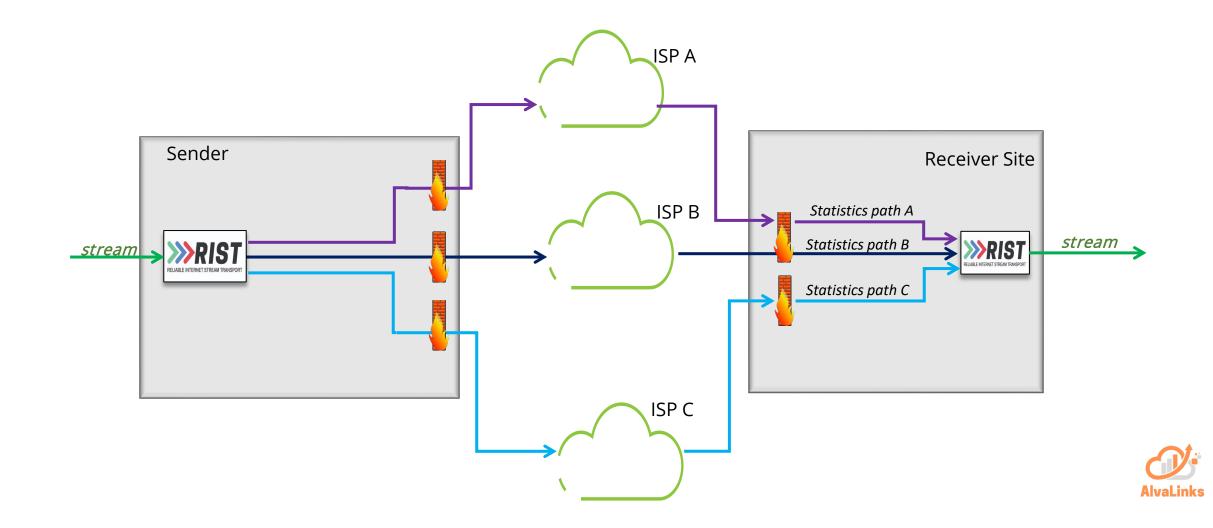
- Links behavior may change during the event/day and over time
- Standard Error recovery is not designed to overcome link capacity errors
- Some links may be have limited error recovery capacity and respreading of the stream between more available links is more desirable
- Sometimes all we are left with is to command the source to change its compression/components to allow service continuation
- We strive to the highest quality and not to drop to the lowest possible



RIST simple/main profile Load share setup (IP showcase

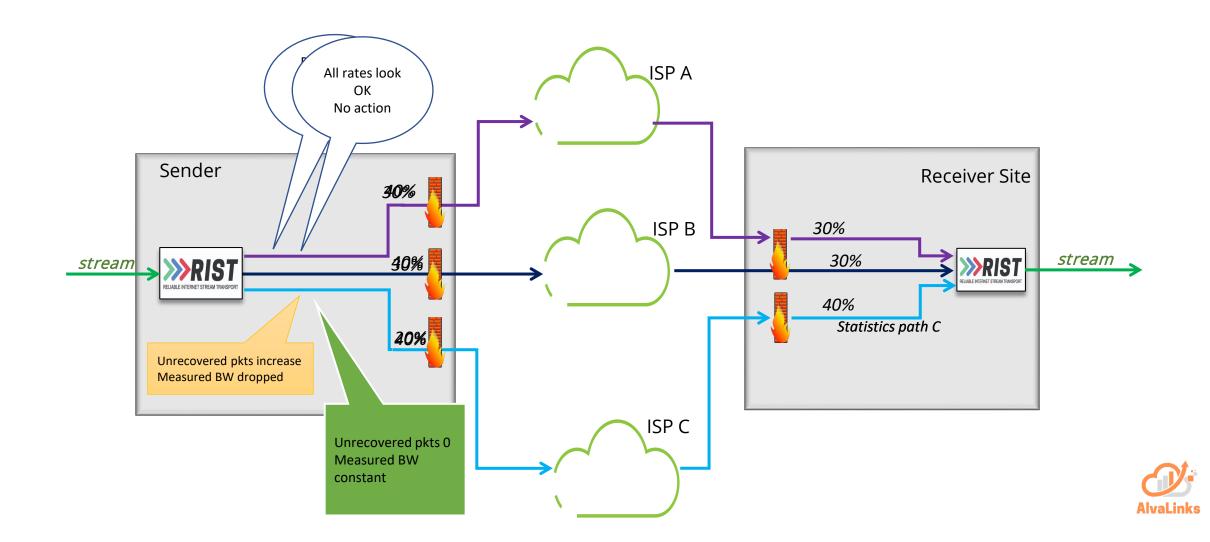


Introduction of Receiver Statistics per link (IP SHOWCASE

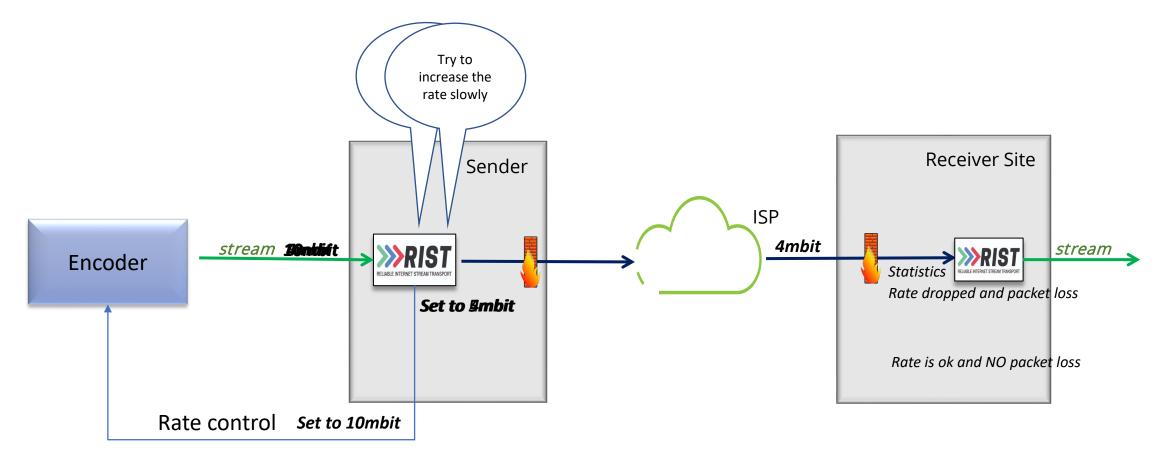


Lets see an example for multi link











What else can I do?



- Select the best available resource (WAN, Wifi, 5G, Starlink)
- Dynamically change the stream partition between different paths
- Some ideas:
 - Stream partition

Send the high priority services over the most reliable path

- Send the others on the other paths
- Component partition
 - Send the Video Stream to the cloud using the lowest delay/error path
 - Send the Audio and auxiliary data to the cloud over a higher delay link that may need error correction

Use of receiver statistics for before and during the transmission. This Will provide better results and over all continuity of service





- Now you have more information at your hand.
- Receiver reports coupled with RTT and Jitter information give a broad view for your IT organization and your network provider.
- You can get more visibility on how the data is traveling and received at the destination to allow you to react and change the behavior.
- A smart implementation will take the handle and do an auto pilot for you, to guaranty the best performance and delay.

Any Questions?

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