SRT
SECURE
RELIABLE
TRANSPORT

Secure Reliable Transport Protocol

Deployment Guide

Version 1.1 I Issue 01

HVS-ID-DG-SRT-1.1

SRTALLIANCE

www.srtalliance.org

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Appendix A: Additional Information

Copyright

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About This Guide

Welcome to the Deployment Guide. This guide describes the SRT protocol and how it is configured and used between all supported devices.

About the SRT Alliance

The SRT Alliance, founded by Haivision and Wowza, is a commercially funded group dedicated to managing and supporting the open source implementation of SRT, a transport protocol for enabling the delivery of high-quality, low-latency video across the public Internet. This alliance is accelerating interoperability of video streaming solutions and fostering collaboration with industry leaders to achieve lower latency internet video transport. The SRT Alliance is open to new members. For companies who want to participate actively in growing the ecosystem of SRT in low latency video streaming workflows, please contact us at info@srtalliance.org.

About Haivision

Haivision is a global leader in delivering advanced video networking, digital signage, and IP video distribution solutions. Haivision offers complete end-to-end technology for video, graphics, and metadata to help customers to build, manage, and distribute their media content to users throughout an organization or across the Internet. Haivision has specific expertise in the enterprise, education, medical/healthcare, and federal/military markets.

Haivision is based in Montreal and Chicago, with technical centers in Beaverton, Oregon; Austin, Texas; and Hamburg, Germany.

About Wowza

Wowza Media Systems™ is the recognized gold standard of streaming, that enables organizations to expand their reach and more deeply engage their audiences on any device, anywhere in the world.

Document Conventions

The following document conventions are used throughout this guide.

TIP The light bulb symbol highlights suggestions or helpful hints.

NOTE Indicates a note, containing special instructions or information that may apply only in special cases.

IMPORTANT Indicates an emphasized note. It provides information that you should be particularly aware of in order to complete a task and that should not be disregarded. IMPORTANT is typically used to prevent loss of data.

CAUTION Indicates a potentially hazardous situation which, if not avoided, may result in damage to data or equipment, or minor to moderate injury. It may also be used to alert against unsafe practices.

Introduction to SRT

This document provides guidance on setting up and deploying SRT technology, which is a feature of an increasing number of products.

For general information on creating and managing streams, please refer to the specific documentation for your product(s).

Overview

Secure Reliable Transport (SRT) is a transport technology that optimizes streaming performance across unpredictable networks, such as the Internet.

Secure	*Encrypts video streams*
Reliable	*Recovers from severe packet loss*
Transport	*Dynamically adapts to changing network conditions*

SRT is applied to contribution and distribution endpoints as part of a video stream workflow to deliver the best quality and lowest latency video at all times.

As audio/video packets are streamed from a **source** to a **destination** device, SRT detects and adapts to the real-time network conditions between the two endpoints. SRT helps compensate for jitter and bandwidth fluctuations due to congestion over noisy networks, such as the Internet. Its error recovery mechanism minimizes the packet loss typical of Internet connections. And SRT supports AES encryption for end-to-end security, keeping your streams safe from prying eyes.

SRT Solutions

SRT is a point-to-point, connection-oriented protocol. Each SRT stream is characterized by the transport of one-way multimedia data and bi-directional control messages, with only one UDP-based connection used per SRT stream. Nonetheless, it is possible to configure SRT solutions that encompass a variety of situations.

Point to Point & Interactive

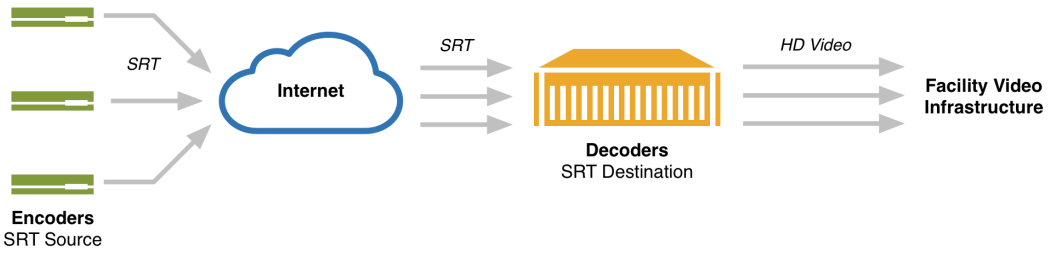
SRT can be easily provisioned for straightforward connections within and between facilities to achieve a low latency, low cost video distribution.

It can be useful over noisy, unreliable local LANs. In large organizations with many users on the same LAN, congestion and packet drops are common — video is very sensitive to this. Even inside the same building on a controlled LAN, SRT can enhance the experience.

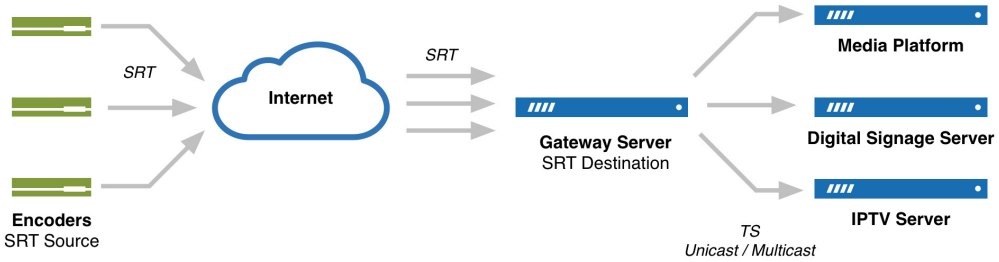
Some networks have VLANs with reserved bandwidth (where routers prioritize traffic) that require a deep knowledge of routers and switches. SRT removes the need for IT intervention to get your video through the network.

The low latency achievable with SRT is even suitable for interactive applications over the Internet.

Contribution & Aggregation

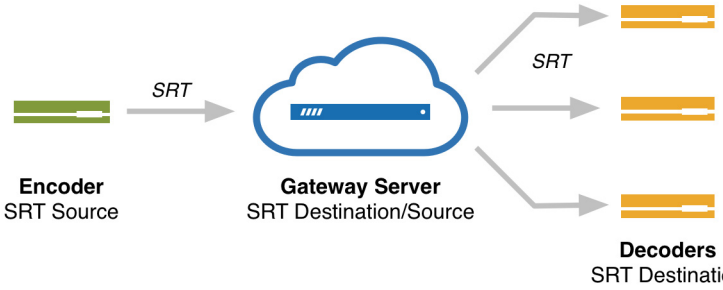


Multiple SRT **source** and **destination** devices can be configured to feed high-demand video infrastructures.



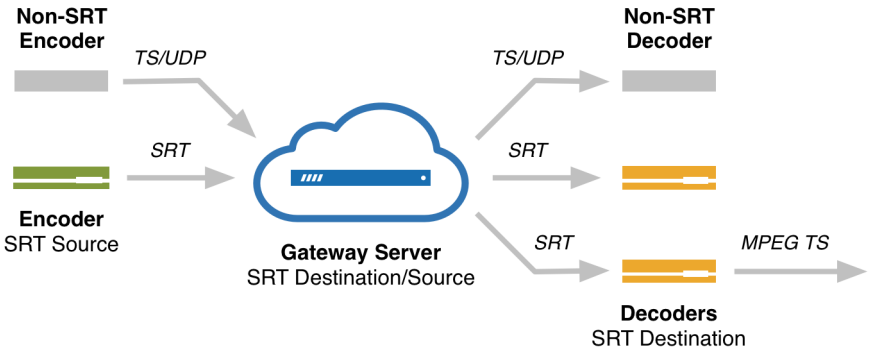
Multiple SRT streams can be aggregated and re-distributed via unicast and multicast into recording, IPTV and digital signage services.

Point to Multi-Point



While SRT operates on point-to-point connections, some gateways are designed to support multiple such connections. A single incoming SRT stream can de redistributed via such gateways to multiple SRT **destination** devices.

SRT Stream Flipping



SRT **destination** devices support "stream flipping" — the ability to convert back and forth between SRT streams and standard MPEG Transport Stream (TS). This is accomplished by de-encapsulating an incoming SRT stream and re-streaming it as a TS/UDP stream (and vice versa), and is typically done to allow devices (on a local LAN) that do not support SRT to have access to incoming SRT stream content.

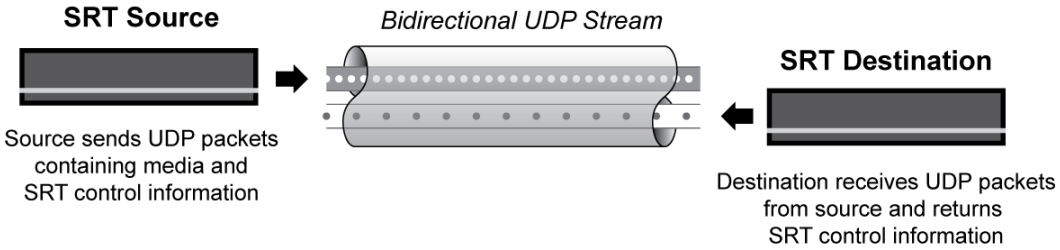
SRT Version History & Compatibility

SRT is designed to be backwards compatible, so that new or upgraded products will be able to support SRT interactions with older ones. Note, however, that newer versions of SRT may have features unavailable on older ones.

Basic SRT Concepts

Source & Destination

The SRT protocol relies on bi-directional UDP traffic to optimize video streaming over public networks. In addition to the video data that is sent in one direction — from a content **source** device to a **destination** — there is a constant exchange of control information between the two endpoints, including "keep alive" packets (if needed) approximately every 10 ms, which enable SRT streams to be automatically restored after a connection loss.



NOTE It is important to understand that with an SRT stream, the **source** is the device that is *sending* the content (audio and/or video data), while the **destination** is the device *receiving* the content. Elsewhere, you may encounter references to an *SRT sender* and an *SRT receiver*, but to avoid confusion in this document we will be using the terms **source** and **destination**.

In some cases, a device can act as both the **source** and the **destination**. For example, a gateway server may act as a **destination** while receiving an SRT stream from an encoder, and then become a **source** device as it re-streams to a decoder.

SRT Call Modes

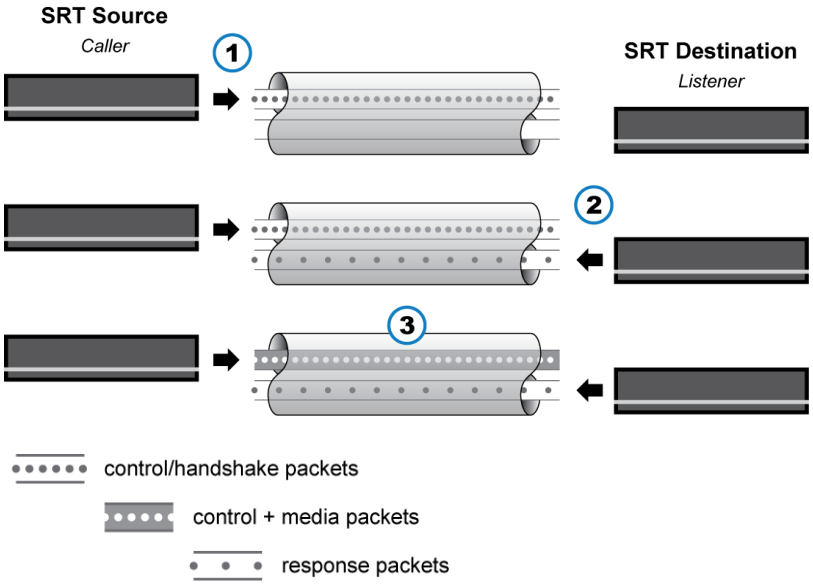
In order to establish a bidirectional stream, SRT employs a handshake mechanism where each device identifies itself as a **Caller** or as a **Listener**. In certain cases, two devices can simultaneously negotiate an SRT session in what is referred to as **Rendezvous** mode. As you are configuring SRT streams, you should understand these handshaking modes and when to apply them:

SRT Call Mode	What it does	When to use it
Caller	Sets a **source** or **destination** device as the initiator of an SRT streaming session. The Caller device must know the public IP address and port number of the Listener.	(1) To initiate point-to-point streaming (e.g. set an encoder to Caller mode to stream to a decoder over a private network, or vice versa). (2) On a **source** or **destination** device that is behind a firewall; may require a network administrator to configure the firewall settings. (3) On a **source** or **destination** device that is not behind a firewall. (4) On a **source** or **destination** device with a dynamic IP address (e.g. a portable encoder using DHCP).
Listener	Sets a device to wait for a request to start an SRT streaming session. The Listener device only needs to know that it should listen for an SRT stream on a certain port.	(1) To participate in a point-to-point streaming session initiated by a Caller (e.g. set a decoder to Listener mode to receive an SRT stream from an encoder). (2) On a **source** or **destination** device that is behind a firewall over which you have control and can open a port. (3) On a **source** or **destination** device that is not behind a firewall, or exposed directly on the Internet. (4) You know that another device will initiate the session.
Rendezvous	Allows two devices to negotiate an SRT session over a mutually agreed upon port. Both **source** and **destination** must be in Rendezvous mode.	(1) When one or both devices are behind firewalls. Once certain settings are in place on the firewall, SRT sessions can be initiated without further intervention by a network administrator.

SRT Call Mode Examples

In Figure 1 below, the SRT **source** device is in *Caller* mode, and the SRT **destination** device is in *Listener* mode. The SRT **source** (Caller) initiates the handshake by sending a series of control packets in a UDP stream (**1**). When the SRT **destination** (Listener) receives these control packets, it responds by sending its own (**2**). Once the handshake has successfully completed, the SRT **source** device begins adding media packets to the UDP stream (**3**).

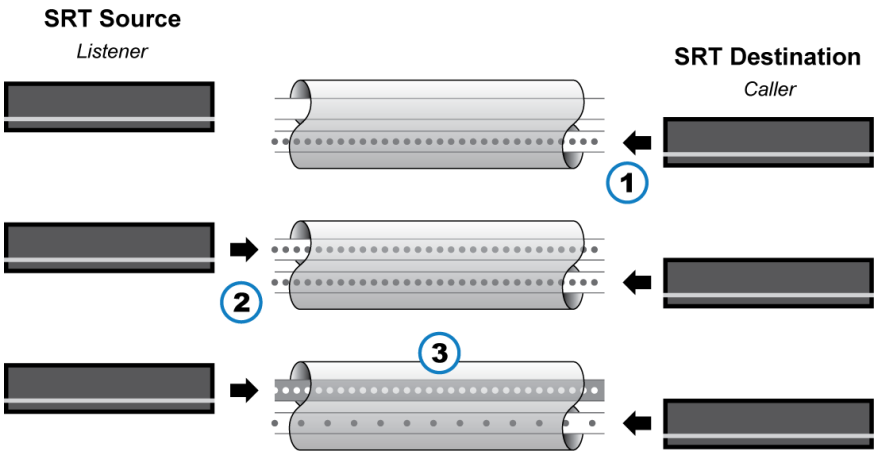
Figure 1 SRT source device initiates connection



NOTE Regardless of which device is in which mode, when the SRT handshake is completed both **source** and **destination** continue to exchange control packets containing information about network conditions, dropped packets, etc. Once communication is established, the notion of which device is Caller and which is Listener becomes unimportant. What matters is the **source**/**destination** relationship, which is decoupled from the caller/listener relationship.

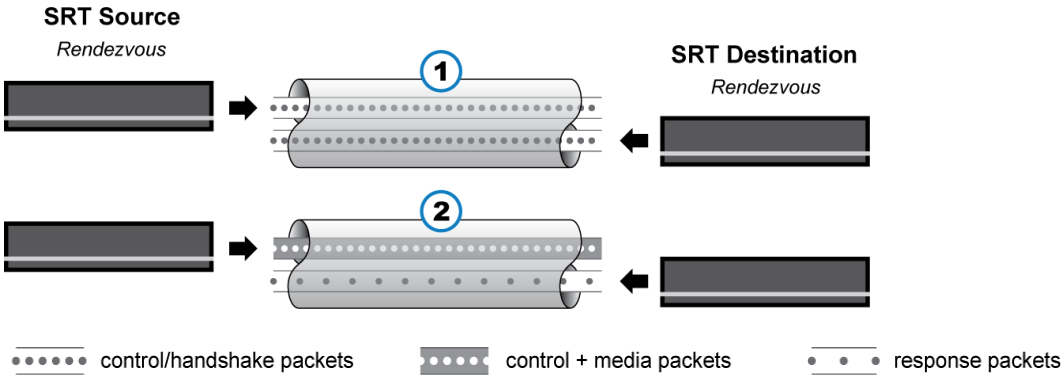
In Figure 2 below, the roles are reversed — the SRT **source** device is in *Listener* mode, and the SRT **destination** device is in *Caller* mode. The SRT **destination** (Caller) initiates the handshake by sending a series of control packets in a UDP stream (**1**). When the SRT **source** (Listener) receives these control packets, it responds by sending its own (**2**). Once the handshake has successfully completed, the SRT **source** device begins adding media packets to the UDP stream (**3**).

Figure 2 SRT destination device initiates connection



In Figure 3 below, both the SRT **source** device and the SRT **destination** device are in *Rendezvous* mode. Both devices send a series of control packets in a UDP stream (**1**). Once the handshake has successfully completed, the SRT **source** device begins adding media packets to the UDP stream (**2**).

Figure 3 SRT connection initiated using Rendezvous



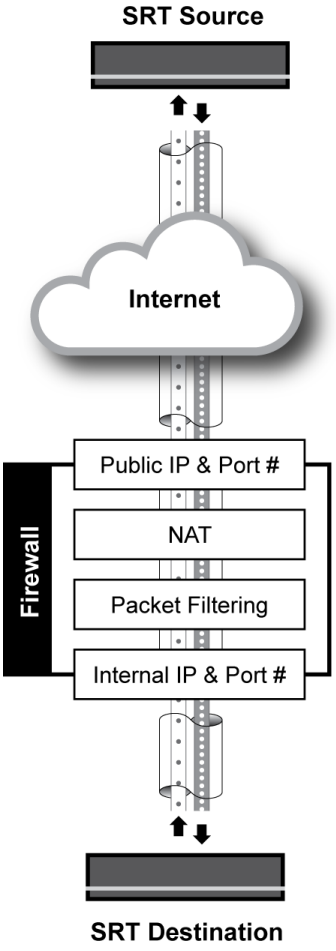
SRT and Firewalls

In many real world situations, particularly those involving transmission over the Internet, SRT streams will have to pass through a firewall at the **source**, at the **destination**, or at both ends. In order to allow this, a network administrator may have to configure certain settings on the firewall(s), specifically those for Network Address Translation (NAT) and packet filtering. The settings will differ depending on whether the devices behind firewalls are in *Caller*, *Listener*, or *Rendezvous* mode.

Example 1

The figure at right illustrates a simple example, where an SRT **source** device is attempting to stream across the Internet to an SRT **destination** behind a firewall. If we consider the case where the SRT **source** device is in *Caller* mode, and the **destination** device is in *Listener* mode, then in order for the handshaking process described earlier to be successfully completed (and an SRT streaming session established) certain conditions must be met:

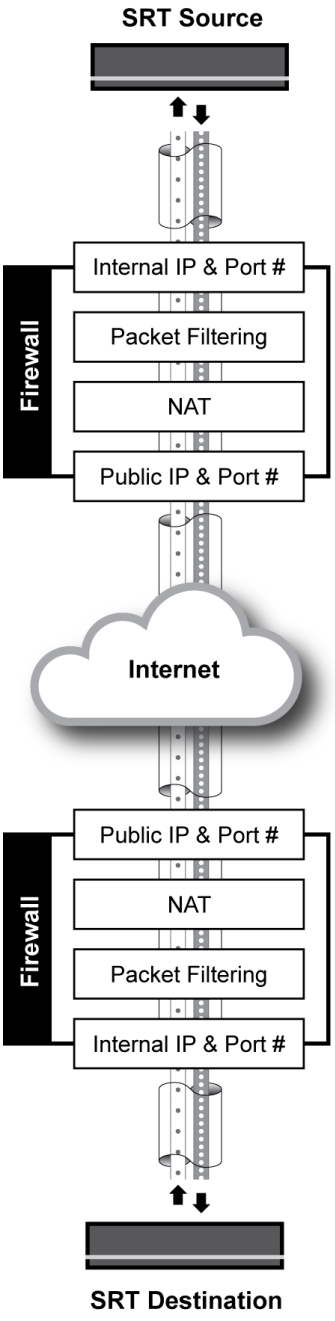
- The SRT **source** device must "know" the public IP address of the firewall, and the port number on which the SRT **destination** device is "listening".
- The firewall must allow the specific **destination** port used by SRT to be accessible from the Internet.
- The firewall must allow bi-directional UDP traffic.
- Port forwarding must be enabled on the firewall to allow data to flow to the IP address and port of the SRT **destination** device.
- Packet filtering must be disabled (to allow the SRT packets to pass through).



Example 2

This figure illustrates a more complex example, where an SRT **source** device behind a firewall is attempting to stream across the Internet to an SRT **destination**, also behind a firewall. If we consider the case where the SRT **source** device is in *Caller* mode, and the **destination** device is in *Listener* mode, then in order for the hand-shaking process described earlier to be successfully completed (and an SRT streaming session to be established) certain conditions must be met:

- The SRT **source** device must "know" the public IP address of the **destination** firewall, and port number on which the **destination** device is "listening". This information usually comes from the IT Admin responsible for the firewall.
- Both firewalls must allow bi-directional UDP traffic.
- Port forwarding (NAT) must be configured on both firewalls to allow data to flow between the SRT **source** and **destination** devices.
- Packet filtering must be disabled on both firewalls (i.e. the SRT packets exchanged between the **source** device and the **destination** device must be allowed to pass through).



NOTE The order in which a firewall performs Network Address Translation and packet filtering will have an impact on how the packet filtering rules are configured.

Deployment Scenarios

This section describes three common scenarios for deploying SRT:

- Scenario 1: Streaming over a private LAN and/or WAN, with no firewall
- Scenario 2: Streaming over a public network with both the **source** and **destination** behind a firewall (Caller/Listener mode)
- Scenario 3: Streaming over a public network with both the **source** and **destination** behind a firewall (Rendezvous mode)

In real-world applications, elements of these scenarios can be employed in various ways.

NOTE Some of the example values provided in these scenarios (IP addresses, port numbers, etc.) are for illustrative purposes only.

NOTE The SRT open source project does not contain any web UI elements. For illustrative purposes, this section contains examples of a web interface based on existing SRT-enabled products.

Scenario 1: Streaming over a LAN or private WAN

In this scenario, an SRT **source** device (the *Caller*) initiates a point-to-point session with an SRT **destination** device (the *Listener*) over a LAN (Local Area Network) or a private WAN (Wide Area Network).

Step 1 — Configure the encoder

On the encoder (the SRT **source** device), do the following:

1. Using the settings in the table below, create and start an Output Stream on the encoder (the SRT **source** device):

Setting	Example	Description
Protocol	TS over SRT	SRT is based on the UDP protocol.
Mode	Caller	Encoder will initiate the SRT session.
Address	198.168.2.20	The target address for the SRT stream, which is the IP address of the decoder.
Source Port	20000	The UDP **source** port for the SRT stream, which is the unique port over which the encoder will be sending the SRT stream. You can (optionally) specify the UDP source port. If not filled in, an ephemeral source port will be assigned (between 32768 and 61000).
Destination Port	30000	The port over which the decoder will be listening.

Step 2 — Configure the Decoder

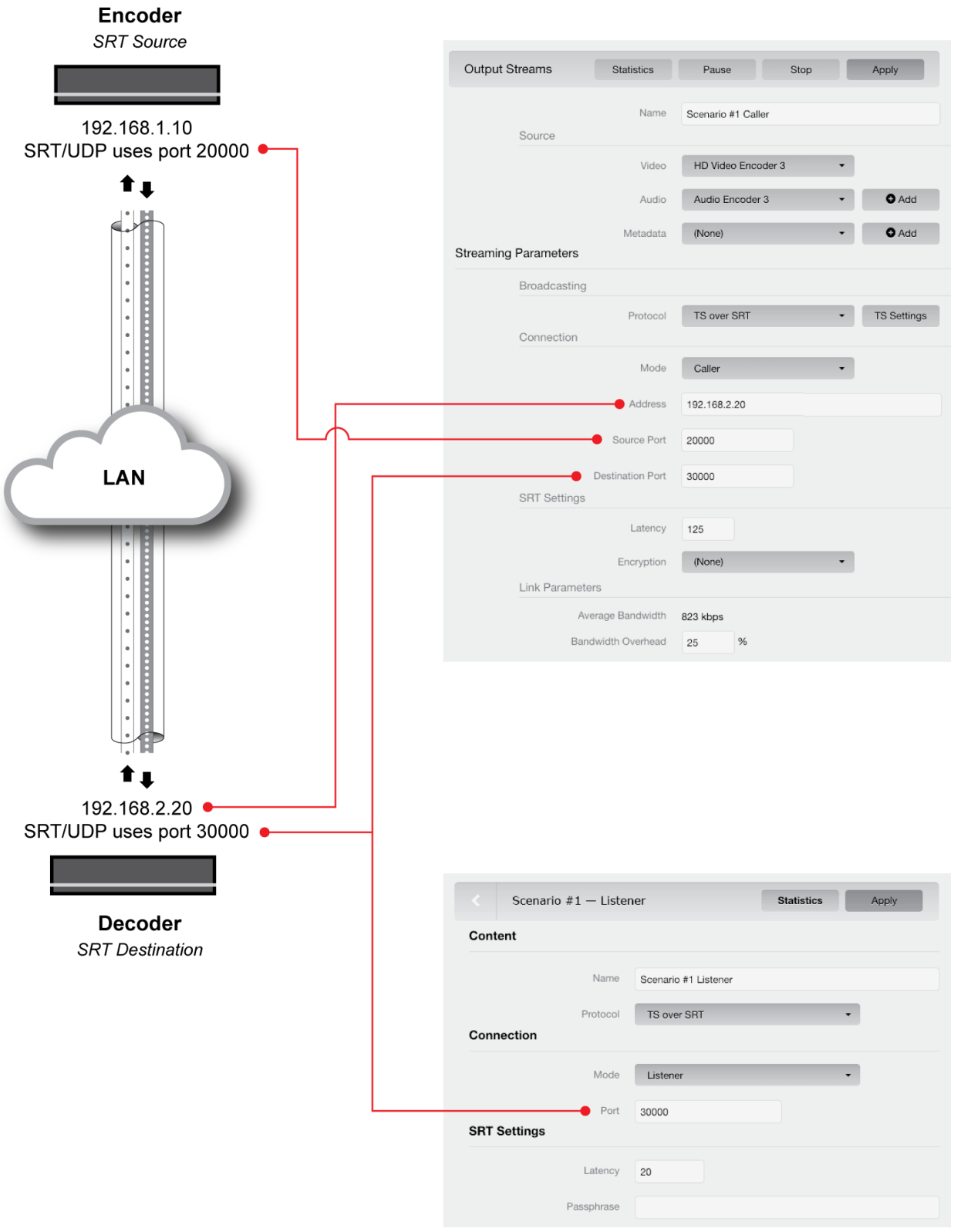
On the decoder (the SRT **destination** device), do the following:

1. Using the settings in the table below, create an Input Stream on the decoder (the SRT **destination** device):

Setting	Example	Description
Mode	Listener	The decoder will wait for the **source** device to initiate the SRT session.
Destination Port	30000	This is the port on which the decoder will be listening (the port to which **Firewall B** will be forwarding SRT packets).

The encoder and decoder will handshake and establish an SRT session. The encoder will send the video stream to the decoder, which will process the stream and return control packets that include congestion data, latency and other statistics. The encoder will use this information to adapt its transmission (resend lost packets, adjust bit rate, etc.).

Note that when the SRT handshake is completed both source and destination continue to exchange control packets. Once communication is established, the Caller or Listener designation becomes unimportant. What matters is the source/destination relationship, or video flow, which is decoupled from the caller/listener relationship.


Encoder
SRT Source
192.168.1.10
SRT/UDP uses port 20000
LAN
192.168.2.20
SRT/UDP uses port 30000
Decoder
SRT Destination
Output Streams
Statistics
Pause
Stop
Apply
Name
Scenario #1 Caller
Source
Video
HD Video Encoder 3
Audio
Audio Encoder 3
Add
Metadata
(None)
Add
Streaming Parameters
Broadcasting
Protocol
TS over SRT
TS Settings
Connection
Mode
Caller
Address
192.168.2.20
Source Port
20000
Destination Port
30000
SRT Settings
Latency
125
Encryption
(None)
Link Parameters
Average Bandwidth
823 kbps
Bandwidth Overhead
25
%
Scenario #1 — Listener
Statistics
Apply
Content
Name
Scenario #1 Listener
Protocol
TS over SRT
Connection
Mode
Listener
Port
30000
SRT Settings
Latency
20
Passphrase

Scenario 2: Streaming using Caller & Listener modes

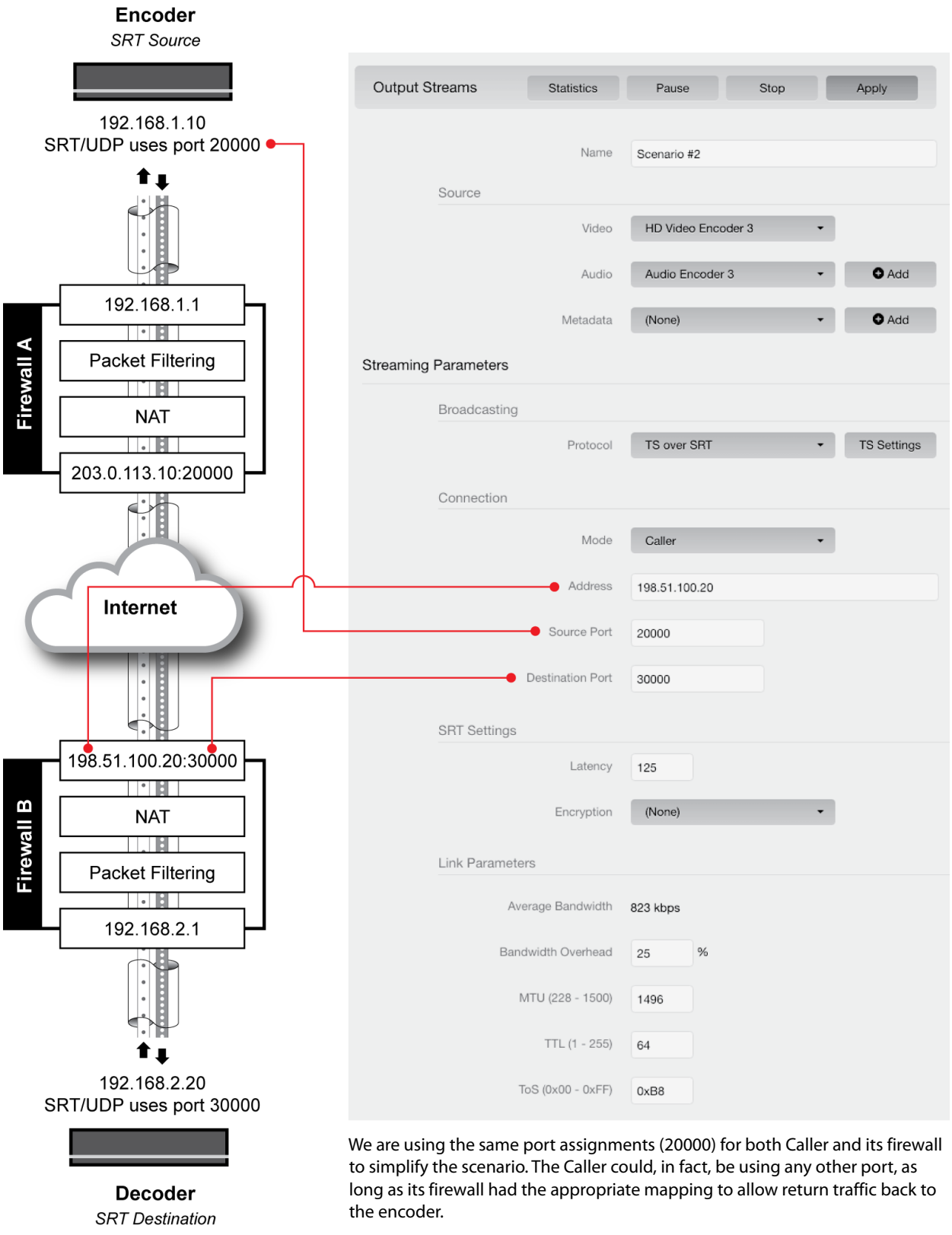
In this scenario, an SRT **source** device (an encoder in *Caller* mode) behind a firewall initiates a point-to-point session over the Internet with an SRT **destination** device (a decoder in *Listener* mode), also behind a firewall.

Step 1 — Configure the Encoder

On the encoder (the SRT **source** device), do the following:

1. Using the settings in the table below, create and start an Output Stream on the encoder (the SRT **source** device):

Setting	Example	Description
Protocol	TS over SRT	SRT is based on the UDP protocol.
Mode	Caller	Encoder will initiate the SRT session.
Address	198.51.100.20	The target address for the SRT stream, which is the public IP address of **Firewall B** (at the **destination**).
Source Port	20000	The unique UDP **source** port for the SRT stream; you can leave the default value (Auto-assign), in which case an ephemeral port is assigned, or, if required by your organization's IT policies, enter a specific (static) port number. If you use Auto-assign, then **Firewall A** must be configured to map ANY **source** port from its local side to a specific port on its public side so that return traffic can be directed to the encoder.
Destination Port	30000	The port over which the decoder will be listening. This is the publicly mapped port number for the SRT **destination** device (i.e. the port **Firewall B** opens for the SRT session).This port must be known (it can't be "any").



We are using the same port assignments (20000) for both Caller and its firewall to simplify the scenario. The Caller could, in fact, be using any other port, as long as its firewall had the appropriate mapping to allow return traffic back to the encoder.

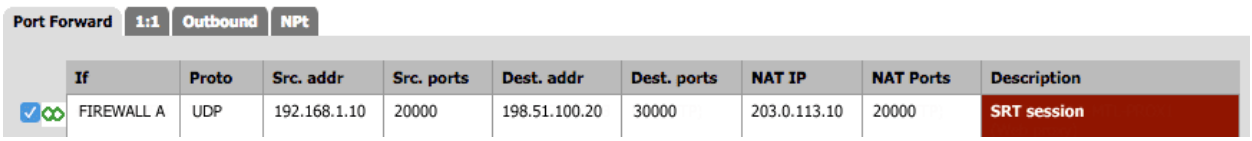
Step 2 — Configure the firewall at the source

On **Firewall A** (at the **source**), do the following:

1. Using the settings in the table below, create an outbound NAT rule that allows bi-directional UDP traffic, with a port forwarding entry for incoming traffic on the firewall's public IP address/port that forwards it to the encoder's IP address/port number:

Setting	Example	Description
Protocol	UDP	SRT is based on the UDP protocol.
Source IP	192.168.1.10	The encoder's IP address
Source Port	20000	In this case, we are using a static port assignment. for the **source** port, but if auto-assigned it can be anything within the ephemeral port range (typically 32768 to 61000 on Linux devices).
Destination IP	198.51.100.20	This is the public IP address of **Firewall B**
Destination Port	30000	This is the port over which the decoder will be listening.
Outbound NAT Source Port	20000	Your firewall must support Outbound NAT Source Port (which disables outbound NAT port rewrite).Otherwise, Rendezvous mode may be required (see Scenario #3).

Here is an example of an outbound NAT rule for Firewall A:



Step 3 — Configure the firewall at the destination

On **Firewall B** (at the **destination**), do the following:

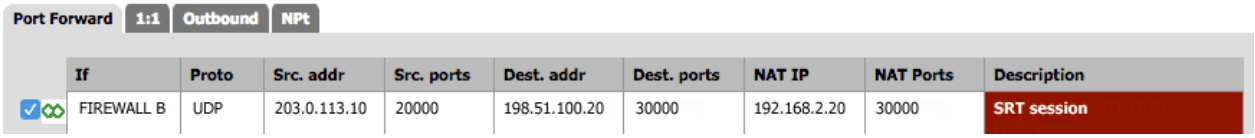
1. Using the settings in the table below, create an inbound NAT rule to enable forwarding of SRT traffic to the decoder's IP address/port number:

Setting	Example	Description
Protocol	UDP	SRT is based on the UDP protocol.
Source IP	203.0.113.10	This is the public IP address of the firewall at the **source** (**Firewall A**).
Source Port	20000	This must match the Outbound NAT Source Port on the firewall at the **source** (**Firewall A**).

Setting	Example	Description
Destination IP	198.51.100.20	This is the public (external) IP address of the firewall at the **destination** (**Firewall B**).
Destination Port	30000	This is the public (external) port of the firewall at the **destination** (**Firewall B**), which in this example is also the port over which the decoder will be listening (dstport).
Redirect Target IP	192.168.2.20	This is the address of the decoder (the internal destination IP).
Redirect Target Port	30000	This is the port over which the decoder will be listening (the internal destination port, or dstport).

NOTE The Redirect Target IP and Port are the internal IP and port number that the destination device is using. Destination IP and Port are the firewall's external interface. The port numbers don't have to be the same. For clarity, it may be useful to always use the same port number, but this is up to your Firewall Administrator to decide.

Here is an example of an inbound NAT rule for Firewall B:

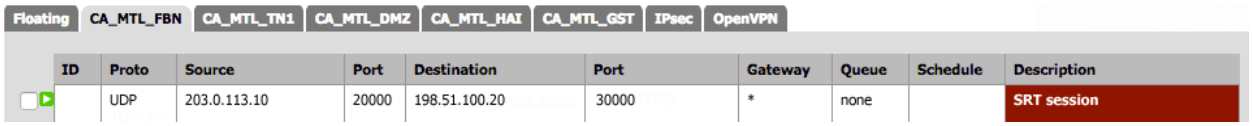


2. Using the settings in the table below, create a packet filtering rule to allow SRT packets to pass freely to and from the decoder's IP address/port number:

Setting	Example	Description
Protocol	UDP	SRT is based on the UDP protocol.
Source IP	203.0.113.10	This is the public IP address of the firewall at the **source**.
Source Port	20000	This must match the Outbound NAT Source Port on the firewall at the **source**.
Destination IP	192.168.2.1 or 198.51.100.20	Depending on your firewall, the NAT rules may be applied before or after the packet filtering rules. This will impact the filtering rule definition. If the NAT is applied before, you have to specify the Firewall B internal IP address. If the NAT is applied after, you have to specify the Firewall B public IP address.

Setting	Example	Description
Destination Port	30000	This is the port over which the decoder will be listening (dstport).
Policy	Accept/Pass	Allows packets to pass freely between **source** and **destination**.

Here is an example of packet filtering rule for Firewall B:



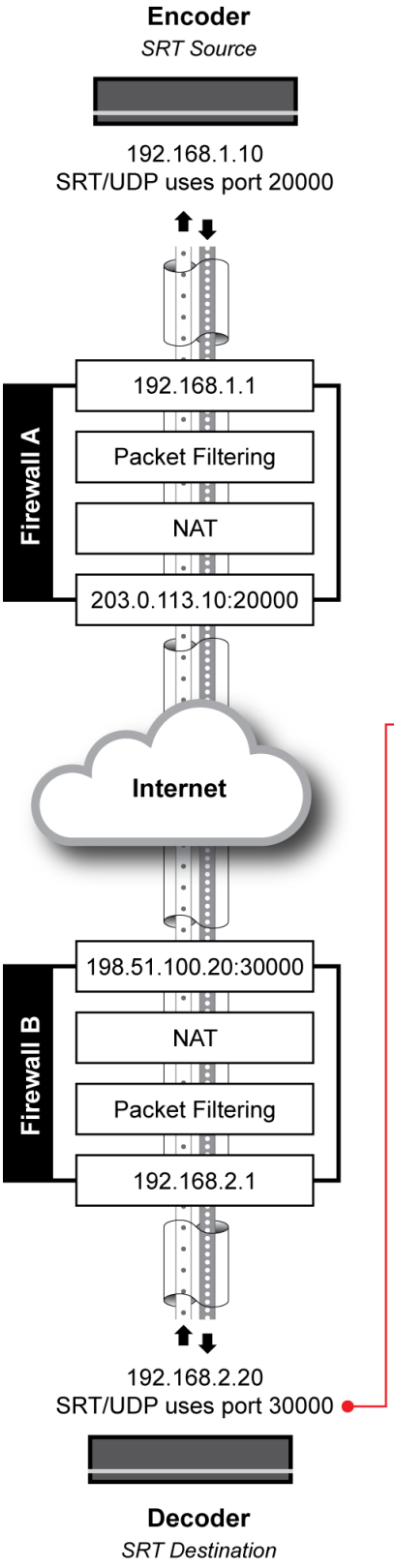
Step 4 — Configure the Decoder

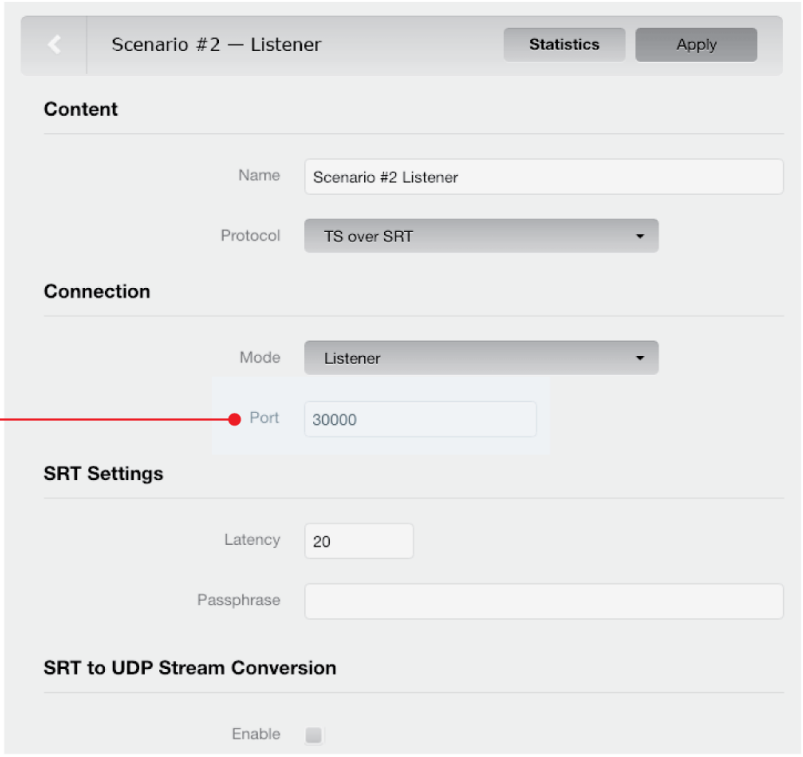
On the decoder (the SRT **destination** device), do the following:

1. Using the settings in the table below, create an Input Stream on the decoder (the SRT **destination** device):

Setting	Example	Description
Mode	Listener	The decoder will wait for the **source** device to initiate the SRT session.
Destination Port	30000	This is the port on which the decoder will be listening (the port to which **Firewall B** will be forwarding SRT packets). If you have a NAT translation rule on Firewall B, the Destination Port is the port to which the rule will be forwarding packets.l

Once all settings have been applied, the encoder and decoder will handshake and establish an SRT session. The encoder will send the video stream to the decoder, which will process the stream and return control packets that include congestion data, latency and other statistics. The encoder will use this information to adapt its transmission (resend lost packets, adjust bit rate, etc.).





We are using the same port assignments (30000) for both Listener and its firewall to simplify the scenario. The Listener could, in fact, be using any other port, as long as its firewall had the appropriate mapping (i.e. how to redirect traffic received on port 30000 to the decoder).

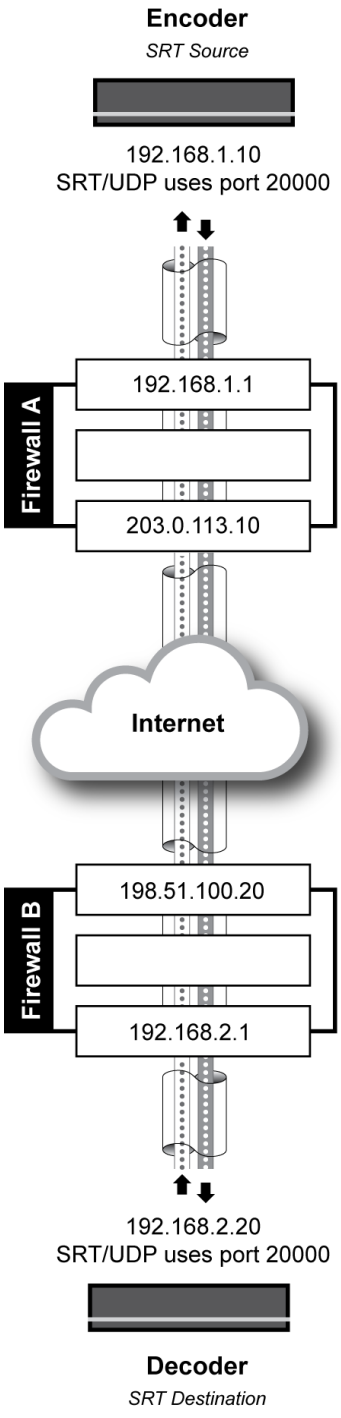
Firewall Notes

- If the **source** device's port is auto-assigned, the firewall at the **source** must have an outbound NAT rule for [**source** port] set to "any".
- If the **source** device's port is specified, then the same value should be used in the outbound NAT rule.
- If a **destination** firewall has a filtering rule that matches a **source** port with a **source** IP, you must disable outbound port rewrite on the **source** firewall. Disabling this option allows the **source** firewall to map any port from the SRT **source** device to a unique, pre-defined port after the NAT rules have been applied.
- Depending on your firewall, the NAT rules may be applied before or after the packet filtering rules. This will affect the filtering rule definition. If the NAT rules are applied before, you have to specify the firewall's internal IP address. If the NAT rules are applied after, you have to specify the firewall's public IP address.

IMPORTANT Point-to-point sessions through firewalls can be done in reverse, with the SRT **source** device in *Listener* mode and the SRT **destination** device in *Caller* mode.

Scenario 3: Streaming using Rendezvous mode

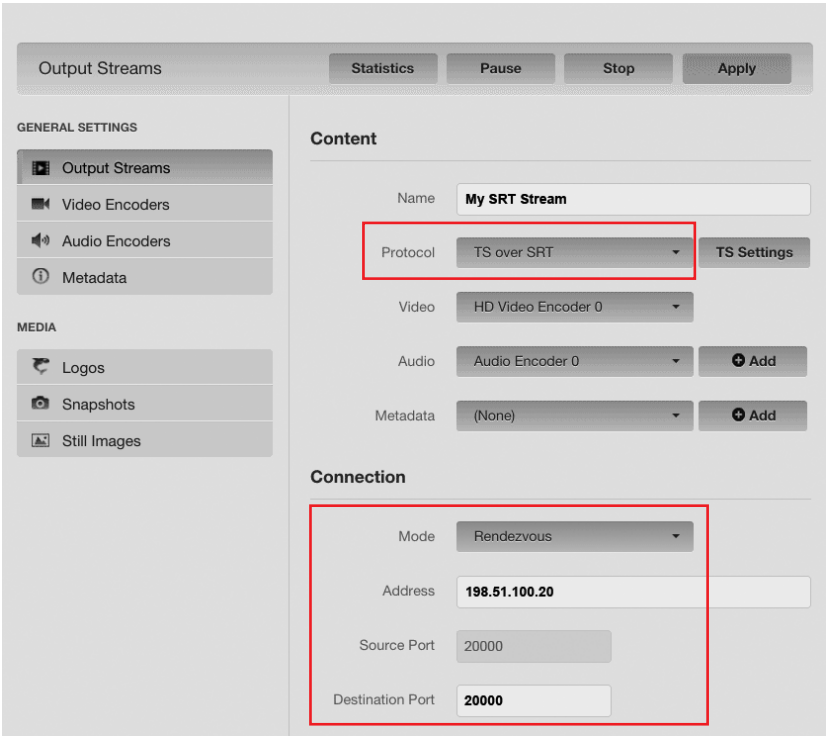
In this scenario, an encoder in *Rendezvous* mode behind a firewall and a decoder, also in *Rendezvous* mode and behind a firewall, mutually establish a point-to-point SRT streaming session over the Internet.



Step 1 — Configure the Encoder

1. Using the settings in the table below, create and start an Output Stream on the encoder (the SRT **source** device):

Setting	Example	Description
Protocol	TS over SRT	SRT is based on the UDP protocol.
Mode	Rendezvous	Encoder will attempt to initiate the SRT session, and listen for incoming SRT connection requests.
Address	198.51.100.20	This is the target address for the SRT stream, which is the public IP address of the firewall at the **destination**.
Source Port	20000	For Rendezvous mode, **source** and **destination** ports are the same.
Destination Port	20000	This is the port over which the decoder will be listening



Step 2 — Configure the firewall(s)

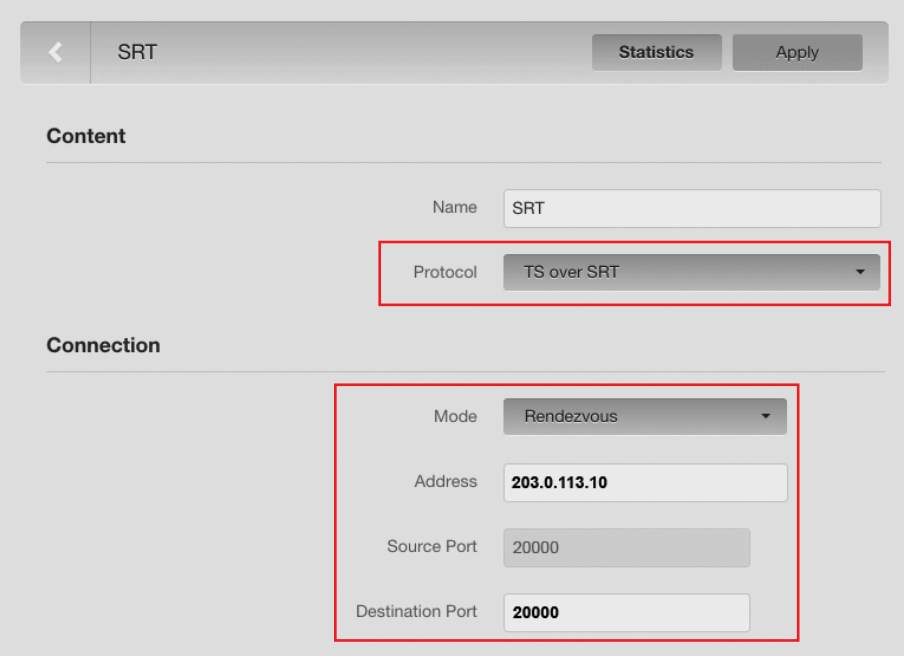
- Make sure that the firewalls between the **source** and **destination** devices have port rewriting turned off (i.e. static port mapping must be allowed). In this scenario, for example, this is necessary to allow the encoder and decoder to establish an SRT session over 20000.

Step 3 — Configure the Decoder

On the decoder (the SRT **destination** device), do the following:

1. Using the settings in the table below, create an Input Stream on the decoder (the SRT **destination** device):

Setting	Example	Description
Protocol	TS over SRT	SRT is based on the UDP protocol.
Mode	Rendezvous	Decoder will attempt to initiate the SRT session, and listen for incoming SRT connection requests.
Address	203.0.113.10	This is the public IP address of the firewall at the **source**.
Source Port	20000	In Rendezvous mode, the **source** port must be the same as the **destination** port.
Destination Port	20000	This is the port over which the encoder will be listening.



Once all settings have been applied, the encoder and decoder will handshake and establish an SRT session. The encoder will send the video stream to the decoder, which will process the stream and return control packets that include congestion data, latency and other statistics. The encoder will use this information to adapt its transmission (resend lost packets, adjust bit rate, etc.).

Rendezvous Mode and Firewalls

Rendezvous mode allows SRT traffic between the **source** and **destination** to traverse a firewall without the need for an IT administrator to open a port, as long as the firewalls are "stateful".

Even if there is no rule to explicitly allow an SRT **destination** device to communicate with the outside world, its control packets will nonetheless be able to return to the SRT **source** device because of the "connection tracking" feature of stateful firewalls. *Rendezvous* mode uses this behavior to create "holes" through both firewalls.

Connection Tracking

Stateful firewalls maintain a connection tracking table, which is dynamically built based on actual traffic passing through the firewall.

In a connection tracking table a typical entry might consist of UDP traffic from a device with a **source** IP and port number, which is converted by NAT, and then connected to the public IP of a firewall on that **destination** port:

Protocol	Source → Router → Destination
UDP	192.168.1.10:20000 --> 203.0.113.10:20000 --> 198.51.100.20:20000

If you have a call coming in from the other endpoint, then you would see the same entry in reverse in the connection tracking table:

Protocol	Source → Router → Destination
UDP	198.51.100.20:20000 --> 203.0.113.10:20000 --> 192.168.1.10:20000

Typically, an encoder would start an SRT session in *Caller* mode, with a decoder in *Listener* mode waiting for control packets. In *Rendezvous*, both are sending control packets to initiate the session. The outgoing packets create an entry in the table. When the incoming packets arrive, they create a complementary entry. This tricks the firewall into thinking that the inbound packets are the responses to the outbound ones, and so it permits the packets to pass through for the duration of the streaming session.

Rendezvous mode allows the **source** and **destination** devices to "punch out" holes from the inside of their respective firewalls. The only conditions are that both devices must be in *Rendezvous* mode, both must be using the same port number, and there must be an outbound entry set up on each firewall so that the **source** port number is preserved (i.e. so that there is no need to create input entry rules).

Other Considerations

- It is important that the port number used is unique. Using a firewall's option to accept **any** (ephemeral) port may be problematic in a scenario where there are multiple encoders sending out streams.
- If the firewall at the **source** is set to allow the **source** device to use **any** (ephemeral) port, then outbound port re-writing must be disabled for *Rendezvous* mode. The port assignment must be static so the SRT participants can meet.
- If either the SRT **source** or **destination** device is behind a firewall that does not allow outgoing traffic, you will need to configure the firewall to allow outgoing traffic.
- There are still occasions where it would be necessary (or better) to manually configure *Caller* and *Listener* settings to traverse firewalls at each end. For example, in high security environments that employ "egress filtering", firewalls are often configured to prevent arbitrary outbound ports from being assigned. An internal address cannot send to a random outbound port. In such cases, both the **source** and **destination** devices would be registered at the firewall, which would block everything else.

Configuring SRT Streams

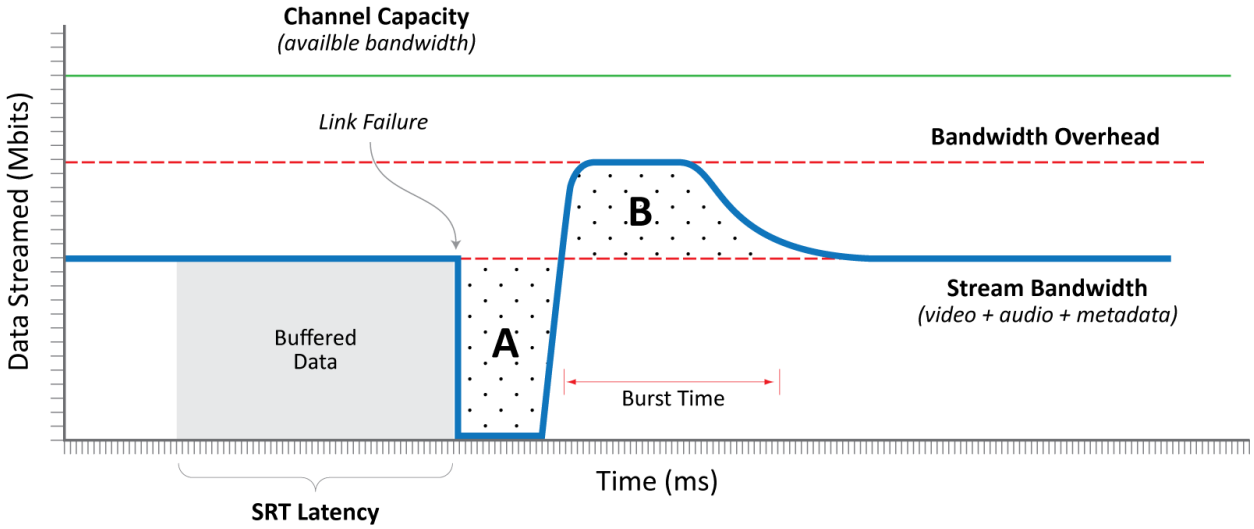
NOTE This section describes how to configure and tune an SRT stream. For complete details on how to configure a stream, please refer to the User's Guide for your device.

The SRT open source project does not contain any web UI elements. For illustrative purposes, this section contains screen shots from Haivsion's SRT-enabled products.

Background

Along with the standard parameters associated with any streaming output, there are other important values you must specify for an SRT stream. To introduce you to these parameters, let's take a look at a hypothetical example of an SRT stream being sent to a destination device, and see what happens over time.

The diagram below depicts a stream being sent over a channel of some kind, such as a LAN or Internet connection, with a certain capacity. Packets being sent from a source are stored at the destination in a buffer. Let's say at some point there is a total link failure, and then, shortly after, the connection is re-established. So, for a short period of time, the destination is receiving no data.



SRT deals with such situations in two ways. The first is that the destination relies on its buffer to maintain the stream output at its end. The size of this buffer is determined by the

SRT Latency setting. Once the link is re-established, the source is able to resume sending packets, including re-sending the packets lost during the link failure. To handle this extra "burst" of packets, an SRT stream allows for a certain amount of overhead. This Bandwidth Overhead is calculated such that, in a worst case scenario, the source can deliver the number of packets "lost" during the link failure (area A) over a "burst time" (area B), where area B must be equal to area A. The maximum time period for which a burst of lost packets can be sustained without causing an artifact is:

SRT Latency (ms) * Bandwidth Overhead (%) ÷ 100

Configuring an SRT Stream

With your source and destination devices set up (including having established call modes and any firewall settings), follow these steps to configure an SRT stream:

1. Measure the **Round Trip Time (RTT)** using the ping command.
   - If ping does not work or is not available, set up a test SRT stream and use the RTT value from the Statistics page.
   - If the RTT is <= 20 ms, then use 20 ms for the RTT value. This is because SRT does not respond to events on time scales shorter than 20 ms.
2. Measure the **Packet Loss Rate**.
   - A channel's Packet Loss Rate drives the SRT Latency and Bandwidth Overhead calculations. This loss rate can be extracted from iperf stats.
     - If using iperf is not possible, set up a test SRT stream, and then use the resent bytes / sent bytes (as reported on the SRT stream's Statistics page) over a 60 second period to calculate the Packet Loss Rate as follows:

       Packet Loss Rate = Resent Bytes ÷ Sent Bytes * 100
3. Using the following table*, find the **RTT Multiplier** and **Bandwidth Overhead** values that correspond to your measured Packet Loss Rate:

Worst Case Loss Rate (%)	RTT Multiplier	Bandwidth Overhead (%)	Minimum SRT Latency (for RTT <= 20 ms)
<= 1	3	33	60
<= 3	4	25	80
<= 7	5	20	100
<= 10	6	17	120

* This table takes into account *constant loss* and *burst loss*. See "Packet Loss Rate" on page 28 for more information.

4. Determine your SRT Latency value using the following formula:

   SRT Latency = RTT Multiplier * RTT

   - If RTT < 20, use the **Minimum SRT Latency** value in the table above.

5. Measure the nominal Channel Capacity available to the SRT stream using the iperf utility.
   - If iperf does not work or is not available, set up a test SRT stream and use the **Max Bandwidth** or **Path Max Bandwidth** value from the Statistics page.
6. Determine the stream bitrate.
   - The steam bitrate is the sum of the video, audio and metadata essence bit rates, plus an SRT protocol overhead. It has to respect the following constraint:

     Channel Capacity > SRT Stream Bandwidth * (100 + Bandwidth Overhead) ÷ 100
   - If this is not respected, then the video/audio/metadata bitrate must be reduced until it is respected.
   - It is recommended that a significant amount of headroom be added to cushion against varying channel capacity, so a more conservative constraint would be:

     0.75 * Channel Capacity > SRT Stream Bandwidth * (100 + Bandwidth Overhead) ÷ 100
7. Determine if the SRT stream has been set up correctly.
   - The best way to determine this is to set up a test SRT stream and look at the SRT Send Buffer graph on the Statistics page of the source device. The send buffer value should never exceed the SRT Latency bound. If the two plot lines are close, increase the SRT Latency.

SRT Parameters

This section describes the various parameters that have an effect on an SRT stream's performance.

Round Trip Time

Round Trip Time (RTT) is the time it takes for a packet to travel from a **source** to a **destination** and back again. It provides an indication of the distance (indirectly, the number of hops) between endpoints on a network. Between two SRT devices on the same fast switch on a LAN, the RTT should be almost 0. Within the Continental US, RTT over the Internet can vary depending on the link and distance, but can be in to 60 to 100 ms range. Trans-oceanic RTT can be 60-200 ms depending on the route.

RTT is used as a guide when configuring Bandwidth Overhead and Latency. To find the RTT between two devices, you can use the ping command. For example:

```
ping 198.51.100.20
```

Response (RTT = 6.633 ms):

```
56 data bytes 64 bytes from 198.51.100.20: seq=1 ttl=64 time=6.633 ms
```

You can also find the RTT for an active SRT streaming session displayed on the **Statistics** page.

RTT Multiplier

The RTT Multiplier is a value used in the calculation of SRT Latency. It reflects the relationship between the degree of congestion on a network and the Round Trip Time. As network congestion increases, the rate of exchange of SRT control packets (as well as retransmission of lost packets) also increases. Each of these exchanges is limited by the RTT for that channel, and so to compensate, SRT Latency must be increased. The factor that determines this increase is the RTT Multiplier, such that:

```
SRT Latency = RTT Multiplier * RTT
```

The RTT Multiplier, then, is an indication of the maximum number of times SRT will try to resend a packet before giving up.

Packet Loss Rate

Packet Loss Rate is a measure of network congestion, expressed as a percentage of packets lost with respect to packets sent.

- **Constant loss** refers to the condition where a channel is losing packets at a constant rate. In such cases, the SRT overhead is lower bound limited, such that:

  ```
  Minimum Bandwidth Overhead = 1.65 * Packet Loss Rate
  ```

- **Burst loss** refers to the condition where a channel is losing multiple consecutive packets, up to the equivalent of the contents of the SRT latency buffer. In such cases, the SRT overhead is lower bound limited, such that:

  Minimum Bandwidth Overhead = 100 ÷ RTT Multiplier

- Burst losses that last longer than the SRT Latency will result in stream artifacts. SRT Latency should always be set to a value above your worst case burst loss period.

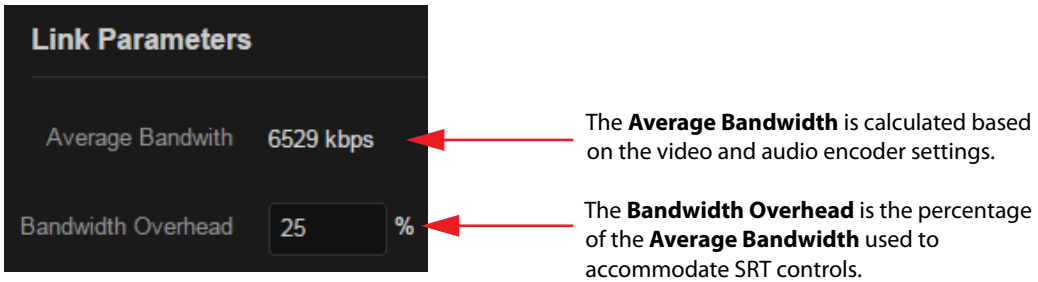
Bandwidth Overhead

The control packets associated with an SRT stream do, of course, take up some of the available bandwidth, as do any media packet retransmissions. When configuring an SRT stream, you will need to specify a Bandwidth Overhead value to allow for this important factor.

The portion of audio and video content in the stream is determined by their respective bit rate settings, which are configured on the audio and video encoders themselves. SRT Bandwidth Overhead is calculated as a percentage of the A/V bit rate, such that the sum of the two represents a threshold bit rate, which is the maximum bandwidth the SRT stream is expected to use.

The SRT Bandwidth Overhead is a percentage you assign, based in part on the quality of the network over which you will be streaming. Noisier networks will require exchanging more control packets, as well as resending media packets, and therefore a higher percentage value.

NOTE SRT Bandwidth Overhead should not exceed 50%. The default value is 25%.



Sample Bandwidth Overhead Calculation

Let's say you are streaming video at a bit rate of 1000 kbps, and audio at 128 kbps. This gives a total of 1128 kbps, which we will round up to 1200 kbps to account for any metadata and other ancillary data. This is the Average Bandwidth, which is calculated automatically based on your actual output settings. If you accept the default Bandwidth Overhead setting of 25%, then the total bandwidth reserved for the SRT stream will be:

1200 + (25% * 1200) = 1500 kbps (1.5 Mbps)

This is the maximum bandwidth SRT will use. If there is no loss, only a slight overhead for control is used. As long as this total SRT bandwidth is less than or equal to the bandwidth available between the SRT **source** and **destination** devices, the stream should flow from one to the other without incident.

Latency

There is a time delay associated with sending packets over a (usually unpredictable) network. Because of this delay, an SRT **source** device has to queue up the packets it sends in a buffer to make sure they are available for transmission and re-transmission. At the other end, an SRT **destination** device has to maintain its own buffer to store the incoming packets (which may come in any order) to make sure it has the right packets in the right sequence for decoding and playback. **SRT Latency** is a fixed value (from 20 to 8000 ms) representing the maximum buffer size available for managing SRT packets.

An SRT **source** device's buffers contain unacknowledged stream packets (those whose reception has not been confirmed by the **destination** device).

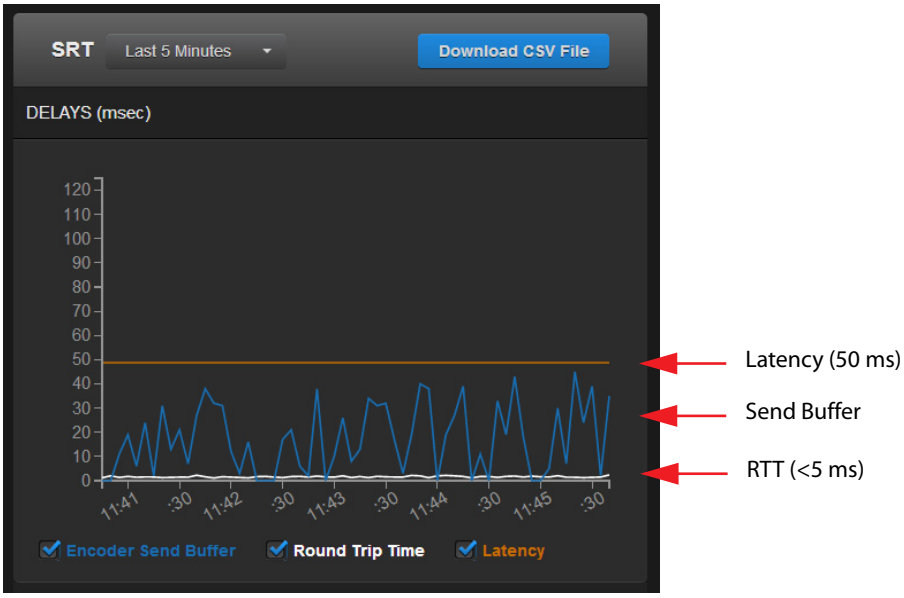
An SRT **destination** device's buffers contain stream packets that have been received and are waiting to be decoded.

The SRT Latency should be set so that the contents of the **source** device buffer (measured in msecs) remain, on average, below that value, while ensuring that the **destination** device buffer never gets close to zero.

The value used for SRT Latency is based on the characteristics of the current link. On a fairly good network (0.1-0.2% loss), a "rule of thumb" for this value would be four times the RTT. In general, the formula for calculating Latency is:

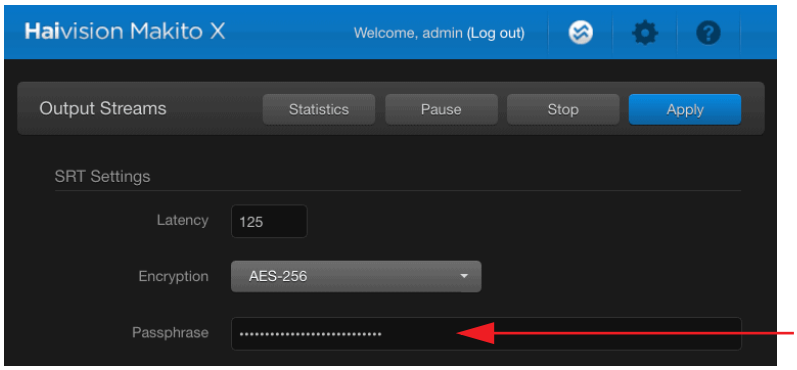
SRT Latency = RTT Multiplier * RTT

SRT Latency can be set on both the SRT **source** and **destination** devices. The higher of the two values is used for the SRT stream.



Encrypting SRT Streams

SRT streams can be encrypted using AES cryptographic algorithms, and decrypted at their **destination**. To implement encryption on an SRT stream, you must specify the type of encryption on the **source** device, and then a pass phrase on both **source** and **destination**.



If encryption is enabled, the **Passphrase** entered on both **source** and **destination** devices must match.

Encryption	This parameter specifies the AES (Advanced Encryption Standard) encryption key length and cipher. Range: AES-128, AES-256
Passphrase	This specifies a string used to generate the AES encryption key wrapper via a one-way function such that the encryption key wrapper used cannot be guessed from knowledge of the password. Range: 10-79 UTF-8 printable characters

Optimizing SRT Performance

SRT **source** and **destination** devices exchange a variety of information about network conditions and packet transfer, which allows them to negotiate the best possible delivery of the main audio/video content.

NOTE The open source SRT project contains all the underlying functionality necessary to create a graphical representation of this exchange, which could then be made available from the web interface of any SRT device.

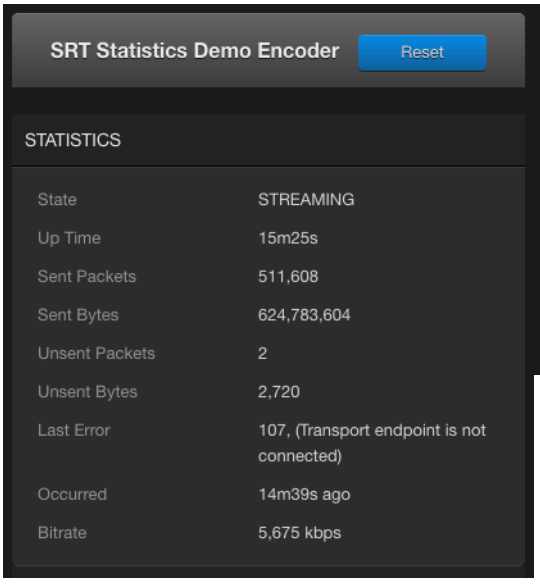
Monitoring and understanding the statistics data can help you to tune and optimize your SRT streaming performance.

For illustrative purposes, what follows are descriptions of the various sections of the **Statistics** page using images from an SRT exchange between a Makito X Encoder (**source**) and Makito X Decoder (**destination**). These are intended to show how, by comparing the related sections between SRT **source** and **destination** devices, you can determine performance characteristics and identify opportunities to optimize the stream.

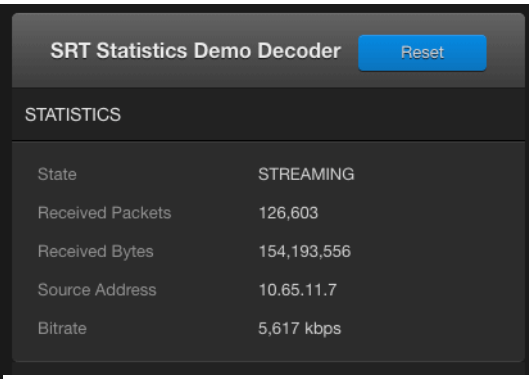
Statistics

The Statistics section provides an overview of an SRT stream's condition: its current state, the packets and bytes sent and received, the current bit rate, and (in the case of the **source** device), how long the stream has been active.

Makito X Encoder (SRT **source** device)



Makito X Decoder (SRT **destination** device)



Src	Dest	Parameter	Description
X		State	The current operating status of the stream (STREAMING, STOPPED, CONNECTING, LISTENING, SECURING, SCRAMBLED, or PAUSED)
X		Up Time	The length of time the stream is actively streaming (e.g., 1d22h5m41s); only available when State is STREAMING.
X		Sent Packets	Number of UDP packets sent for that stream.
X		Sent Bytes	Number of bytes sent for that stream.
X		Unsent Packets	Number of UDP packets not sent for that stream (displayed only when not 0).
X		Unsent Bytes	Number of bytes not sent for that stream (displayed only when not 0).
	X	Received Packets	Number of UDP packets received for that stream.
	X	Received Bytes	Number of bytes received for that stream.
X		Last Error	The last error reported by the SRT subsystem. All errors are recorded in a log file (see "SRT Logs" on page 41).
X		Occurred	The time elapsed since Last Error was reported
	X	Source Address	IP address of the SRT source device
X	X	Bitrate	The stream bitrate (in kbps).
X	X	Reset	Click to reset the **Statistics** page.

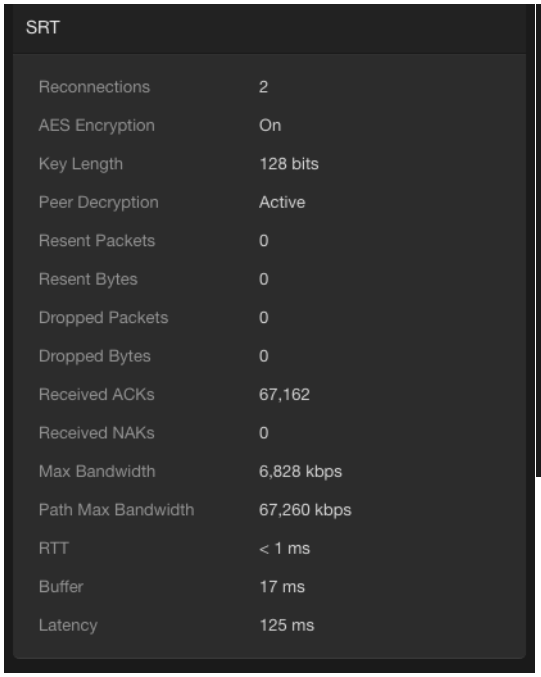
What to look for:

- Make sure the connection state is not SCRAMBLED or STOPPED.

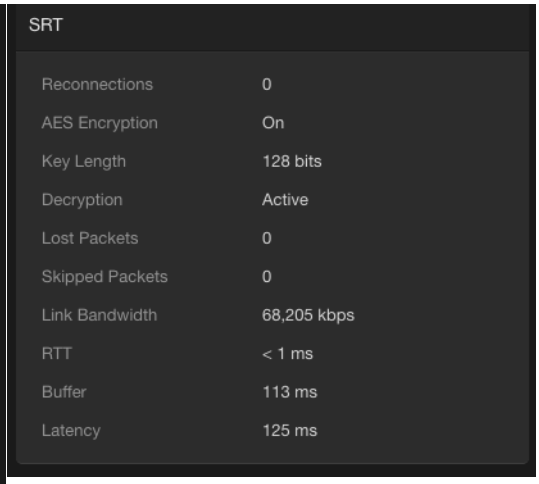
SRT

The SRT section provides a detailed look at the SRT traffic.

Makito X Encoder (SRT **source** device)



Makito X Decoder (SRT **destination** device)



Src	Dest	Parameter	Description
X	X	Reconnections	Number of reconnections since the stream started. Severe network congestion may cause the connection to drop, but it will be automatically reconnected.
X		AES Encryption	Indicates status of AES Encryption (if enabled)
X		Key Length	Indicates the specified key length for AES encryption.
X		Peer Decryption	Indicates whether or not the SRT stream is being successfully decrypted.
	X	Decryption	Indicates whether or not the SRT stream is being successfully decrypted.
X		Resent Packets	Number of packets retransmitted based on reports from the **destination** device (or lack thereof).
X		Resent Bytes	Total bytes retransmitted.
X		Dropped Packets	Number of packets reported missing by the **destination** device. This is the raw number of packets dropped by the network. These packets may be recovered by retransmission by the **source** device, and so do not necessarily result in any video artifacts.

Src	Dest	Parameter	Description
X		Dropped Bytes	Number of dropped bytes.
	X	Lost Packets	Number of packets reported missing by the decoder.
	X	Skipped Packets	Packets that have arrived at the **destination** device too late, or that never arrive at all. If the “time to play” for a packet has passed, and it is either not at the decoder or arrives after the content it is associated with has already played, that packet is reported as “skipped” on the **destination** device. Usually this results in some type of video artifact (a replayed frame or video blocking).
X		Received ACKs	Total number of acknowledgment and feedback packets received from the **destination** device (this is a measure of transmission progress).
X		Received NAKs	Total number of negative acknowledgment packets received from the **destination** device.
	X	Link Bandwidth	Estimated maximum bandwidth available as viewed from the **destination** device.
X		Max Bandwidth	Maximum bandwidth used by the **source** device for this SRT stream (i.e. the current total of audio/video bit rate plus ancillary data plus the SRT Bandwidth Overhead).
X		Path Max Bandwidth	The same value as **Link Bandwidth**. The **destination** device sends the value to the **source** device with an acknowledgment packet.
X	X	RTT	Round Trip Time (RTT) is the time required for a packet to travel from a specific **source** to a specific **destination** and back again.

Src	Dest	Parameter	Description
X	X	Buffer	Size of the SRT buffer (in milliseconds). The SRT **source device's** buffers contain unacknowledged stream packets (those whose reception has not been confirmed by the decoder). The size of the **source** device buffer, in the absence of congestion or packet loss, is around the RTT value. In the presence of recoverable packet loss, the value should be between those for RTT and Latency. The SRT **destination device's** buffer contains stream packets received and waiting to be forwarded or decoded. This statistic shows the portion of the **destination** device's buffer up to the first missing packet (i.e. the time remaining to transmit the missing packet before it's too late). The value of the **destination** device buffer in the absence of packet loss is just below the latency value. In the presence of packet loss, it is between 0 and the latency value. Note that if you change the bit rate from 6 to 2 Mb/s, the byte count will change but a 2 second buffer will remain 2 seconds.
X	X	Latency	A fixed value (from 20 to 8000 ms) representing the maximum buffer size available for managing SRT packets. Values can be entered on both the **source** and **destination** devices. When the handshake occurs at the initiation of an SRT streaming session, the higher of the two values is implemented on both devices.The **destination** default is set to the minimum (20 ms) so that the value can be controlled from the **source**. The default latency value on the **source** is 125 ms.

What to look for:

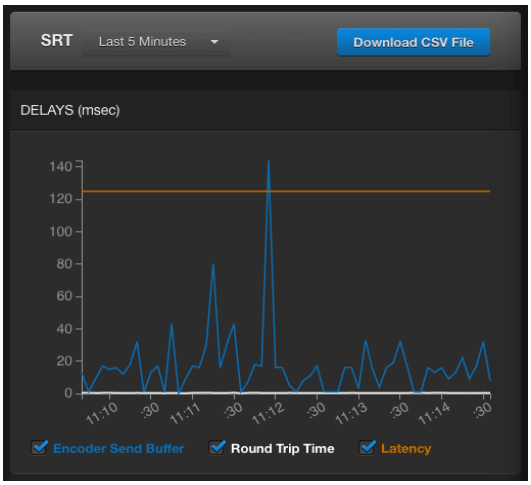
- If you start to see lots of reconnections, this is an indication that there is a problem with the communication channel between devices.
- A steadily increasing number of NAKs indicates a problem.
- It's normal to see a few dropped packets, but there should not be too many. A low number of dropped packets indicates you are using your bandwidth optimally.
- If the number of Skipped Packets increments slowly, the best thing to do is increase the SRT Latency. If it increments in large jumps, the best thing to do is to lower the video bitrate, or to increase the Bandwidth Overhead % if you have available capacity.

TIP When adjusting the SRT parameters, ignore any spikes or other variations that appear on the Statistics charts at the time of the change.

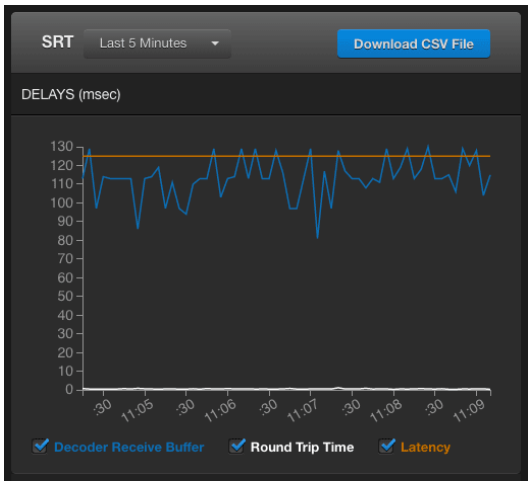
Delays

The Delays section provides a graphical overview of how the SRT stream buffers are handling the traffic.

Makito X Encoder (SRT **source** device)



Makito X Decoder (SRT **destination** device)



Src	Dest	Parameter	Description
X	X	Last X Minutes	Choose a time frame (5 minutes, 60 minutes, or 24 hours) for the data to be displayed in the DELAYS and BANDWIDTH USED graphs.
X	X	Download CSV file	Click to download a CSV file.
X		Encoder Send Buffer	Check this box to view a plot of the contents (in ms) of the encoder buffer over time (blue line in the graph).
	X	Decoder Receive Buffer	Check this box to view a plot of the contents (in ms) of the decoder buffer over time (blue line in the graph).
X	X	Round Trip Time	Check this box to view a plot of the RTT value over time (white line in the graph).
X	X	Latency	Check this box to view a plot of the Latency value over time (orange line in the graph).

What to look for:

- The value for Buffer (blue line) on the source device should normally not exceed the Latency value (orange line). If the Encoder Send Buffer spikes above the latency line, then increase the SRT Latency. You may see occasional spikes (with corresponding anomalies in the display). But if these are few and far between, and the impact on the display of the stream is tolerable, you may choose to ignore these spikes as you tune the stream parameters.
- Ideally, the **destination** device buffer level should be just below the latency value. If the Decoder Receive Buffer goes to 0 often, then there is most likely insufficient BW

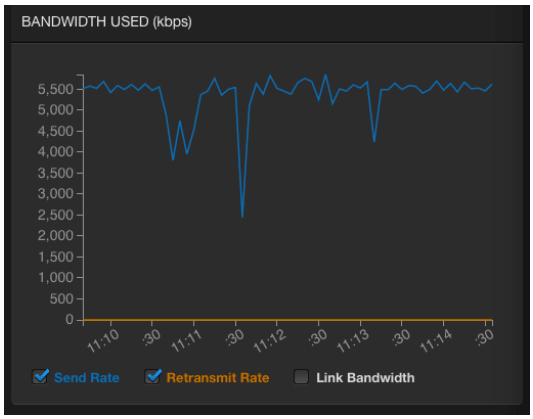
to support the desired bitrate (which should therefore be lowered). If the Decoder Receive Buffer only occasionally goes to 0, then the SRT Latency should be increased.

- On the **source** device, if the Encoder Send Buffer often reaches or exceeds the Latency value, then there is most likely insufficient bandwidth to support the desired bitrate (which should therefore be lowered). If the Encoder Send Buffer only occasionally goes to or above the Latency value, then the SRT Latency should be increased.

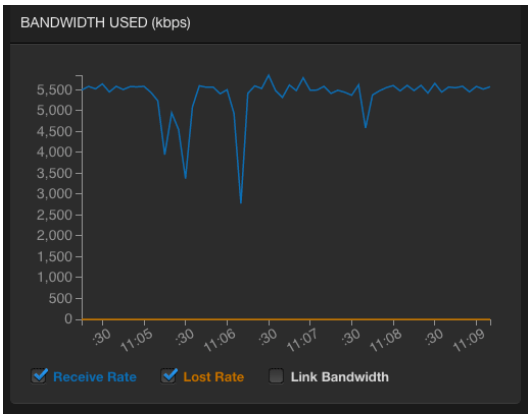
Bandwidth Used

The **Bandwidth Used** section provides a graphical overview of how the network bandwidth available to the SRT stream is actually being used, in terms of the rate at which packets are sent, received, resent and lost, as well as the data volume (rate × time).

Makito X Encoder (SRT **source** device)



Makito X Decoder (SRT **destination** device)



Src	Dest	Parameter	Description
X		Send Rate	Check this box to view a plot of the SRT **source** device's outbound bitrate over time (blue line in the graph).
	X	Receive Rate	Check this box to view a plot of the SRT **destination** device's inbound bitrate over time (blue line in the graph).
X		Retransmit Rate	Check this box to view a plot of the rate at which the SRT **source** device is resending packets (orange line in the graph).
	X	Lost Rate	Check this box to view a plot of the rate at which SRT **source** device is reporting lost packets (orange line in the graph).
X	X	Link Bandwidth	Check this box to view a plot of bandwidth available to the SRT stream (white line in the graph). This is a plot of the Link Bandwidth (or equivalent Max Path Bandwidth on the source device) shown in the SRT panel.

What to look for:

- The use of encryption has an impact on the processing resources of the **source** device, which may have an effect on streaming capacity overall.
- If the SRT **destination** device shows too many skipped or dropped frames, increase the SRT Latency, lower the video bit rate, and/or increase the Bandwidth Overhead %.

Graph Sample Rates

The information presented in the graphs on the **Statistics** page is based on continuous samples taken every 5 seconds over a 24 hour period (sampling does not occur during pauses or disconnects).

The data is displayed in either 60 or 120 columns (depending on the time span being displayed), where each column represents an average value based on the time span being displayed.

Graph Time Span	# Columns (Data Points) Displayed
Last 5 minutes	60
Last 60 minutes	60
Last 24 hours	120

For example, if the graph is showing data from the last 5 minutes (300 seconds), then it will contain 300 ÷ 5 sec. = 60 samples over 60 columns (this works out nicely to one sample per column).

If the graph is showing data from the last 24 hours (86400 seconds), then it will contain 86400 ÷ 5 sec. = 17280 samples over 120 columns. Each column contains the average value of 144 samples (17280 ÷ 120).

SRT Logs

Data associated with an SRT stream is continuously logged (one log per stream). The log file will contain time-stamped entries at 5 second intervals for up to a maximum of 24 hours The sampling is non-contiguous (no log entries are made when the stream is inactive). For example, the log might contain data covering 4 days, but only for 6 hours per day.

From the **Statistics** page, you can download the log file in CSV format, which may be used with applications such as Microsoft Excel or 3rd party log analysis software. It can be useful to examine logs to help with troubleshooting, or to determine how available bandwidth is being used over time, which may allow you to better control associated costs.

The table below shows a sample log with a few rows of data:

DateTime	SendRate	ReSnRate	DropRate	MaxBw	AvailBw	RTT	SendBufs	PdDelay
2016-06-09T10:54:30-0400	3479	0	0	4226	140504	74.417	72	125
2016-06-09T10:54:35-0400	3334	0	0	4226	155224	74.304	77	125
2016-06-09T10:54:40-0400	3385	0	0	4226	92273	74.514	87	125
2016-06-09T10:54:45-0400	3377	0	0	4226	66601	74.248	91	125
2016-06-09T10:54:50-0400	3493	0	0	4226	56608	74.204	81	125
2016-06-09T10:54:55-0400	3358	0	0	4226	156397	74.214	73	125
2016-06-09T10:55:00-0400	3391	0	0	4226	176659	74.338	73	125
2016-06-09T10:55:05-0400	3397	0	0	4226	302024	74.512	70	125
2016-06-09T10:55:10-0400	3460	0	0	4226	72274	74.341	97	125
2016-06-09T10:55:15-0400	3334	0	0	4226	51725	74.264	43	125

APPENDIX A: Additional Information

Frequently Asked Questions

What is the underlying SRT protocol structure?

SRT is based on the User Datagram Protocol (UDP), but has its own mechanisms to ensure real-time delivery of media streams over noisy networks, such as the Internet.

What ports are required to be open through a firewall?

The **source** and **destination** device UDP ports are configurable. Each stream only requires a single port. The port is user-defined and can be between 1025 and 65,535. Specific port requirements for firewalls may depend on the security policies of your organization. Consult with your network system administrator.

What is the bandwidth overhead requirement for a connection traversing the Internet?

BW Overhead specifies the maximum stream bandwidth overhead that can be used for lost packet recovery. The default **BW Overhead** is 25%. Depending on your settings, SRT will either retransmit lost packets quickly (thereby using more bandwidth) or over a longer period (requiring less bandwidth but resulting in higher latency).

How does encryption affect the bandwidth?

Encryption does not affect the bandwidth. However, applying encryption is a processor-intensive task, and may have an impact on the number and bit rate of the streams an encoder is able to output.

Is SRT compatible with wireless networks?

SRT can be used over wireless networks, WiMANs (Wireless Metro Area Networks), LANs, private WANs, or the public Internet.

Does the number of "Lost Packets" increment even when an SRT retransmit is successful? Would lost packets result in visual artifacts or quality issues?

"Lost Packets" and the related "Skipped Packets" are statistics reported by an SRT decoder:

Lost Packets: A hole in the packet sequence has been detected. A request to re-transmit the lost packet has been sent to the **source** device. This lost packet may (or may not) be recovered by the re-transmit request.

Skipped Packets: The time to play the packet has arrived and the lost packet was not recovered, so the decoder will continue playing without it. A video or audio artifact may result from a skipped packet.

How can I determine the SRT bandwidth requirements from a decoder back to an encoder?

SRT back-channel control packets from the decoder to encoder take up a minimum of ~40 Kbps of bandwidth when the channel conditions are perfect. If there are lost packets on the link, then the SRT decoder will generate more signal traffic, proportional to the lost packet rate. A single lost packet will consume about 400 bps of the available bandwidth on the decoder side.

From the decoder back to the encoder, bandwidth usage will increase linearly with the packet loss rate.

How can I determine the bandwidth available between two endpoints before starting an SRT stream?

If you have established an SRT stream, you can view bandwidth information in the statistics from the **source** and **destination** devices.

If no SRT stream is currently running, you can use the Iperf bandwidth measurement utility to get bandwidth and jitter information. You need to specify the port number as well as stop any streaming before using Iperf (to release the ports).

On the **destination** device, make sure any streams are stopped, and then enter the following commands:

```
iperf -s -u -i 1 -p "port#"
```

where "port#" is the same as the port opened on the firewall. The "-u" parameter specifies the use of UDP packets (using TCP would cause the measured available bandwidth value to be lower).

On the **source** device, make sure any streams are stopped, and then enter the following commands:

```
iperf -c "IP ADDRESS OF DESTINATION" -u -b "BW" -i 1 -p "Port#"
```

where "BW" is the appropriate bandwidth in Mbps, and "IP ADDRESS OF DESTINATION" is the IP address of the destination device (use the public facing IP if traversing firewalls).

Example:

```
iperf -c 198.51.100.20 -u -b 5.5m -i 1 -p 20000
```

Result:

```
0.0-10.1 sec 6.50 MBytes 5.41 Mbits/sec 0.247 ms 38/ 4678 (0.81%)
```

where 5.41 Mbits/second is the real bandwidth, 0.247 ms is the jitter, and 0.81% is the percentage of lost packets.

Note that when using Iperf in a UDP client/server configuration, the "server" listens for connections from clients, but it is the "client" that generates the traffic.

Can I stream from one encoder to multiple decoders?

No. SRT is for point-to-point connections. It does not support multicast. If you need to have an SRT stream delivered to multiple decoders, you can use a gateway server.

I'm seeing a "sawtooth" pattern on the Statistics page for my encoder. What does that mean?

It means the decoder is not acknowledging that it has received the packets sent from the encoder. The encoder keeps the packets it has sent in its buffer until it receives a response (this takes at least the equivalent of the round trip time). If the encoder receives acknowledgments promptly from the decoder, the buffer remains relatively empty. Otherwise, the buffer gradually fills until it reaches a point where it must drop the unacknowledged packets, creating the characteristic sawtooth pattern.

This typically occurs when you don't have enough bandwidth to transmit. The buffer value will increase up to the point where it can no longer keep up, and then will drop in a classic sawtooth pattern. In such cases, try increasing the SRT overhead, or lowering your video bit rate.

What does it mean when I see the blue line drop below the white line on the Statistics page for my decoder?

The decoder has a buffer (blue line) that it uses to hold on to what it has received to allow time for retransmission of missing packets. Normally the contents of this buffer (measured in milliseconds) should be between the latency (orange line) configured for the SRT stream, and the round trip time value (white line). If it falls below the RTT, this means the decoder has no packets to play, and not enough time to ask for more. So the video output will be a black screen. It could be that the encoder hasn't given any packets to the decoder, indicating a problem at the source or in the intervening network. But it might just be because there is nothing to display.

When should I start to worry about a rising number of dropped packets?

If the rate is going up constantly, it means that you haven't configured enough bandwidth overhead or latency. Check to see if you are streaming too far above the latency value on the encoder.

How about skipped packets on the decoder?

Sometimes a packet arrives at the decoder ahead of time and sits in the queue, ready to play. But the packet that should be played immediately before arrives late (or never), so the decoder skips the packet sitting in the queue. In other words, the time to play for the associated packet has passed, and it is either not at the decoder or arrives after any associated content has already played. Usually this means some type of video artifact also occurs (a replayed frame or video blocking artifacts).

You could think of a skipped packet as a packet that the decoder drops, except that it doesn't tell the encoder. The decoder sends a positive ACK, even though the packet is "lost" from the point of view of the decoder. It might drop an entire frame because it is corrupted, but the encoder doesn't know it. This is because when things are going badly, the last thing you want to do is to increase the handshake traffic. A packet has a certain time-to-live, and if it doesn't play after that time then it is skipped.

You might see the number of skipped packets increasing on the decoder, without a corresponding effect on the number of dropped packets seen on the encoder. If the number of skipped packets on the decoder increases slowly, you should increase the SRT Latency. If the number of skipped packets on the decoder increments in large jumps, the best thing to do is lower your video bitrate, or increase your Bandwidth Overhead % if you have available capacity.

Do I control the latency value at the source or destination?

You can control the latency, the bit rate, and the overhead percentage on the source device. But you can configure the latency on the destination device as well. The two devices will settle on the maximum value.

How do I decide whether to boost the latency or lower the bit rate?

You have to decide what is most important for you: quality or latency. If low latency is more important to you, then you may see that you are not using your bandwidth as effectively, and image quality may not be optimal. If you want more quality, you need to use the maximum available bandwidth, with minimal overhead, and therefore more latency. An SRT transmission needs either bandwidth or time. So if you are using your maximum bandwidth at a higher bit rate because you want higher quality, you'll have to allow more time to recover from faults.

What happens if I set my latency value too low?

If the delay/latency setting is too low, you will see blue lines (Send Buffer) climbing up past the orange line (Latency) in the graph on the encoder. This will be reflected at the destination as visible artifacts, corresponding to skipped packets on the decoder.

If I see my Send Buffer repeatedly spiking by one or two seconds, by how much should I increase my latency?

In early SRT versions, there was little tolerance. The Send Buffer values had to remain below the Latency to obtain good results. With the most recent SRT version the Send Buffer can occasionally go over one second without being dropped. However, this can cause a problem at the other end, because while the packets are shown as "delivered" on the encoder, they may not arrive at the other end.

Does SRT support Pro-MPEG Forward Error Correction:

No, SRT does not support PRO-MPEG error recovery at this time. Its error recovery mechanism is based on retransmission requests, in the form of NAK (Negative Acknowledgment) packets. The destination device sends NAKs back to the source device

whenever missing packets are detected. The source device responds by re-sending the packets identified in the NAKs.

What are “keep alive” packets?

To maintain an SRT connection, control packets must be sent at an interval of 10 ms (max). When a destination device receives a media packet, it acknowledges the reception by returning an “ack” control packet. If the interval between “ack” packets is greater than 10 ms, “keep alive” packets are sent to keep the connection open.

Note that, unlike with TCP, if an SRT connection is broken, the source device will be unaware that the destination is not receiving packets for up to several seconds before the connection is re-established.

For more information, please visit www.srtalliance.org

Terms and Definitions

The following terms are used in this guide:

Term	Abbreviation	Description
Bandwidth	BW	The data exchange rate (measured in bps, Kbps, or Mbps) of a communications channel, such as an SRT stream.
Bandwidth Overhead		The portion of the total bandwidth of a stream that is required for the exchange of SRT control and recovery packets. This value is usually calculated as a percentage of the base (TS) output bitrate of an encoder (video + audio + metadata + ancillary data). The sum of the base output rate plus the bandwidth overhead determines the maximum bandwidth that can be used by the SRT protocol.
Jitter		Jitter (measured in milliseconds) is the undesired deviation from true periodicity of an assumed periodic signal in electronics and telecommunications, often in relation to a reference clock **source**. In a computer network context, jitter is the variation in latency as measured in the variability over time of the packet latency across a network.
Local Area Network	LAN	A computer network that interconnects computers within a limited area such as an office building.
Latency		The amount of time between the capture and display of a video frame, including the time it takes to encode, transmit and decode. With SRT, latency is a parameter that can be adjusted to increase the buffers at **source** and **destination** to allow those devices to adapt to network conditions.
Network Address Translation	NAT	A functionality that remaps one IP address to another. Network Address Translation allows a device, such as a router or firewall, to redirect traffic between the Internet and a private network.
Packet Loss		Packet loss occurs when data packets fail to reach their **destination**. It can be caused by a number of factors including signal degradation over a network, channel congestion, corruption, faulty networking hardware, faulty network drivers, or normal routing routines (such as DSR, Dynamic Source Routing, in ad hoc networks). Packet loss is measured as a percentage of packets lost with respect to packets sent
Packet Delay Variation	PDV	Packet Delay Variation is the difference in the end-to-end, one-way delay between selected packets in a flow, with any lost packets being ignored. [RFC 3393]
Round-trip Time	RTT	Also called round-trip delay, RTT (measured in milliseconds) is the time required for a packet to travel from a specific **source** to a specific **destination** and back again.

Secure Reliable Transport	SRT	A transport technology developed by Haivision that optimizes streaming performance across unpredictable networks such as the Internet.
Virtual Private Network	VPN	A mechanism for connecting to a private network over a public network like the Internet. Most commonly, a VPN is used to provide a secure, encrypted tunnel through which a remote (authorized) user can access a company network.

About the SRT Alliance

The mission of the SRT Alliance is to support the free availability of the open-source SRT video transport protocol in order to accelerate innovation through collaborative development. Furthermore, the SRT Alliance will promote industry-wide recognition and adoption of SRT as a common and de facto standard for all low latency internet streaming.

One of the important goals of the SRT Alliance is to make new features available to the open-source community, whether submitted for inclusion by community developers, or coming directly from Haivision or Wowza development teams. Community-contributed open source SRT functionality will be available to any developer, and new developments by SRT Alliance members will migrate back into open source SRT on a regular basis.

If you're interested in contributing to this initiative and joining the SRT Alliance, please contact the contact the alliance at info@srtalliance.org.