

SRT IP Input

The Secure Reliable Transport (SRT) input is a new generation IP module. It's an open source video transport protocol optimized for unreliable networks (packet loss, jitter, and fluctuating bandwidth), typically used for transmissions over the Internet. End-to-end 128, 192 or 256 bit AES encryption is available. Different connection modes are provided for easy firewall traversal.

Add Port(s)

Increment IP Increment Ports

SRT Mode	IP Address	Source Port	Destination Port	PSI Mode	Dejitter
<input type="text" value="Search"/> Caller Listener Rendezvous	10.10.110.175	10003	10003	DVB	<input checked="" type="checkbox"/>

A SRT input connection can be configured as three different modes:

Caller	Actively trying to connect to a listener on a specific ip and port.
Listener	Listen for incoming caller on a specific port
Rendezvous	Listen for incoming caller on a specific port


It should be noted that in rendezvous mode, both the destination and source port will always be equal.

Edit Port 5:2

Port	Name	<input type="text"/>
	PSI Mode	DVB
Connection	SRT Mode	Rendezvous
	IP Address	10.10.110.174
	Source Port	10002
	Destination Port	10002
SRT Settings	Encryption Mode	On
	Passphrase	*****
	Receive Latency	120 ms
Dejitter	<input checked="" type="checkbox"/> PCR	<input checked="" type="checkbox"/>
	<input type="checkbox"/> CBR if transparent	<input type="checkbox"/>
	<input type="checkbox"/> Preferred PCR PID	*
	<input type="checkbox"/> Reduced input buffer size	<input type="checkbox"/>

The features presented here are as following:

Mode	Select SRT mode: Caller, Listener or Rendezvous
IP Address	Specify the IP address
Source Port	This will be the port from which SRT will send video data
Destination Port	Select port which SRT should try to connect to
Encryption	Type of encryption: ON or OFF
Passphrase	The password. Minimum length is 10 characters
Receive latency	Set the latency buffer. Gives time to the SRT to do retransmission on lost packets

	<p>The SRT module is limited by the CPU, thus in most cases the bitrate will limit the number of srt connections. Recommended total bitrate for a SRT card is 35 Mbps for both input and output with full 256 AES encryption.</p> <p>Higher bitrate then the recommended value will increase the risk of CC error and triggering of the high CPU load alarm.</p> <p>If total the bitrate is kept bellow the recommended value, then the SRT card supports 32 inputs and 32 outputs.</p>
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SRT IP Output Module

For SRT output modules, the SRT settings tab is shown below:

Edit Settings

Service Components Scrambling Transport **SRT Settings** EMM HbbTV Apps PSI

SRT Settings

SRT Mode: Rendezvous
IPv4/IPv6 Address: 10.10.110.173
Source Port:
Destination Port: 10002

Encryption

Encryption: AES 128
Passphrase: admin12345

Transmission data

Peer latency: 0 ms
Payloadsize packets/frame: 7
Receive latency: 120 ms

Advanced

TTL: 64
Overhead bandwidth: 25 %
TOS: 184

Output Redundancy Activate to enable output redundancy

Apply Cancel OK

SRT Settings:

SRT Mode	Select SRT mode: Caller, Listener or Rendezvous
IP Address	Specify the IP address
Source Port	This will be the port from which SRT will send video data
Destination Port	Select port which SRT should try to connect to

Encryption:

Encryption	Select encryption options: AES128, AES192, AES 256
Passphrase	The password. Minimum length is 10 characters

Transmission Data:

Peer latency	Set the minimum latency for the peer
Receive latency	Set the latency buffer. Gives time to the SRT to do retransmission on lost packets.
Payload size packets/frame	Number of MPEG TS packets in each transmission

Advanced:

TTL	Time to live for IP packet
TOS	Type of IP service
Overhead bandwidth	Specify how much bandwidth above the estimate bandwidth the SRT can use when recovering lost packets