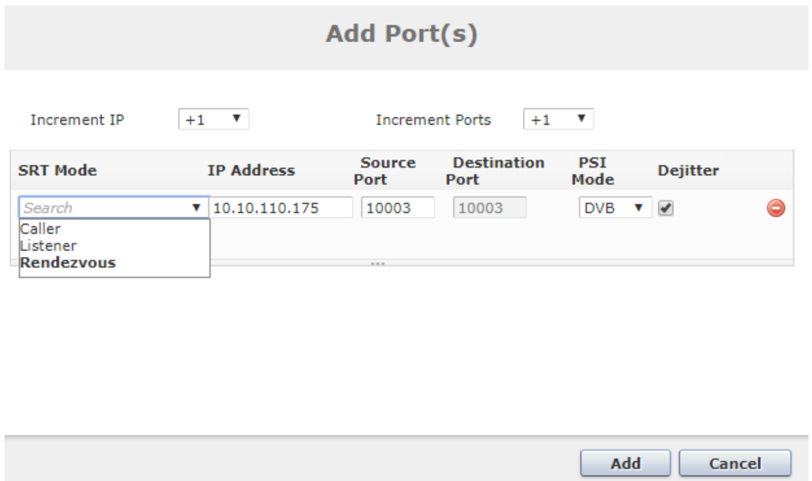
SRT IP Input

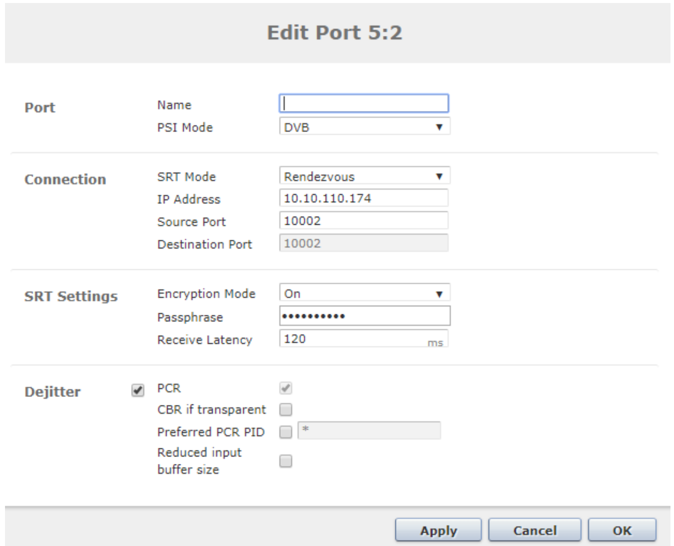
The Secure Reliable Transport (SRT) input is a new generation IP module. It's an open source video transport protocol optimized for unreliable networks (packet loss, jitter, and fluctuating bandwidth), typically used for transmissions over the Internet. End-to-end 128, 192 or 256 bit AES encryption is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



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.



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

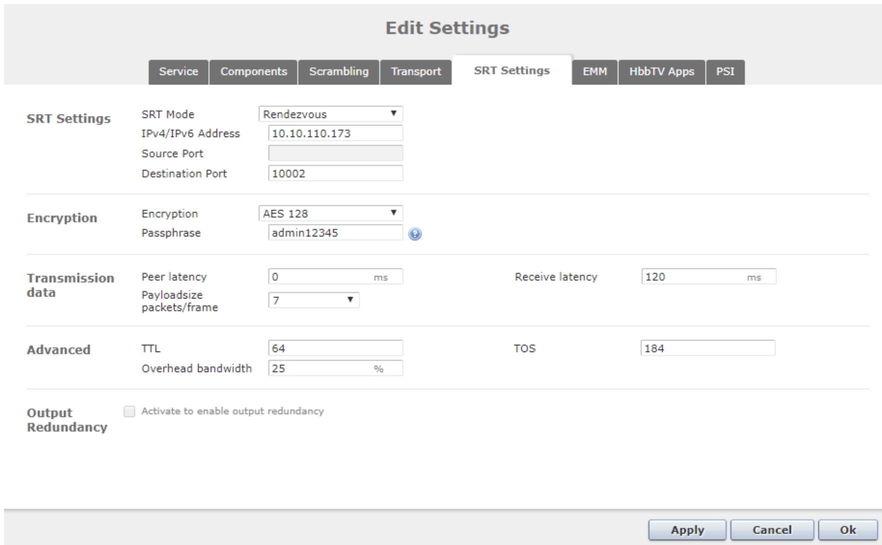
The SRT module is limited by the CPU, thus in most cases the bitrate will limit the number of srt connections. Recommended total bitrate for a SRT card is 35 Mbps for both input and output with full 256 AES encryption.

Higher bitrate then the recommended value will increase the risk of CC error and triggering of the high CPU load alarm.

If total the bitrate is kept bellow the recommended value, then the SRT card supports 32 inputs and 32 outputs.

SRT IP Output Module

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



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