

SRT User Manual

1. SRT introduction

SRT (Secure Reliable Transport) is an open source, free and flexible application specification. It is a transmission technology that can transmit data streams in a complex network environment in real time. It uses the UDP protocol at the transport layer, although the UDP protocol is an unreliable transmission protocol, with the powerful data recovery capability of SRT, coupled with the fast speed and low overhead of the UDP protocol, finally make SRT safe, stable and fast.

2. SRT application

2.1 Point-to-point one-way transmission and video interaction

During the transmission of audio and video streams from the SRT source device (shown as below encoder) to the SRT target device (such as the below decoder), the SRT will detect and adapt to the changing network state between the two devices in real time, resist the bandwidth jitter that caused by network congestion. With its powerful error recovery mechanism, it can minimize the possibility of network packet loss. At the same time, SRT can also perform AES encryption to ensure data security during transmission.



SRT classic transmission way

2.2 Point-to-multipoint transmission

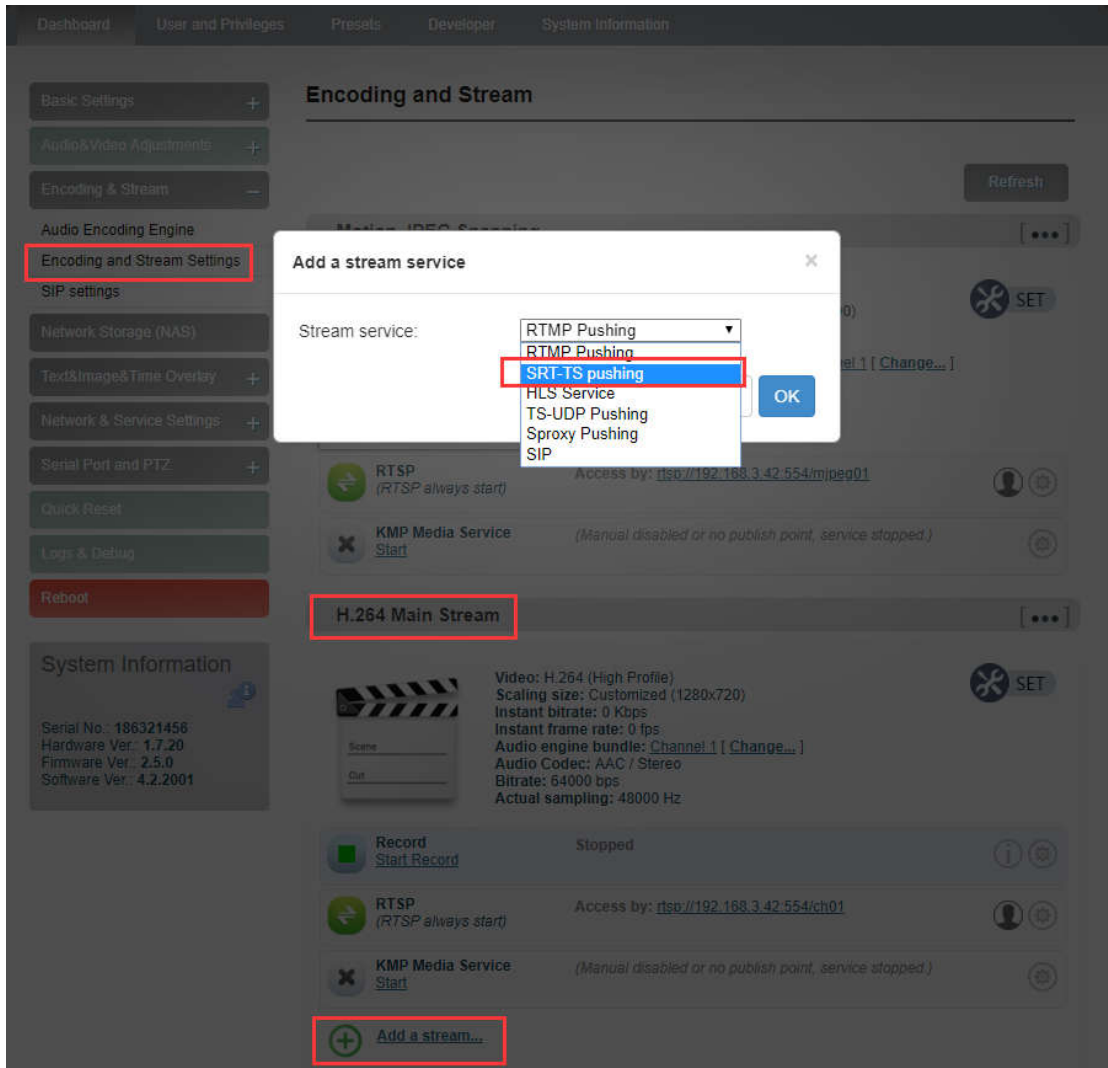
The encoder supports adding multiple SRT points to push directly to multiple target decoder addresses for point-to-multipoint transmission. People can also use a streaming media gateway to distribute the video stream sent by one encoder to multiple decoders. The streaming media gateway is as the central node. The video stream sent by the encoder is received first, and then copied and distributed to multiple decoders for point-to-multipoint video transmission.

2.3 Video streaming protocol conversion and distribution

Media gateway devices (such as Kiloview D300plus/pro) can realize the mutual conversion of various video streaming protocols such as SRT, RTMP and RTSP. It copies, converts and distributes for each video stream, which greatly increases the compatibility of SRT system. It also enables various network flows such as local RTSP and RTMP to smoothly integrate into SRT system, improving the flexibility of video forwarding.

3. Encoder SRT function configuration

3.1 Click “Encoding & Stream”--“Encoding and Stream Settings”--“H.264 main stream”--“Add a stream”. Choose “SRT-TS pushing”, click “OK” and add a SRT service.



3.2 Click the icon  for SRT configuration.



3.3 Fill in the pushing address and port, click “Save”. Then SRT pushing is working.

Primary Stream SRT-TS Pushing Parameters

Open SRT service

Handshake Mode

Address

Port
(Please fill in the value of 0-65536)

Transport Delay(ms)
(Please fill in the value of 0-5000)

Advanced Settings

Encryption mode

AES Key
(Please fill in 10-32 letters or numbers.)

Bandwidth Overhead(%)
(Please fill in the value of 0-100)

Load Size
(Please fill in the value of 0-1456)

Handshake mode: The device set as CALLER mode will act as the caller of SRT session;

Address: Set IP address of the receiving port;

Port: Set the listening port corresponding to the receiving port;

Transport delay: Set it based on the performance of the current network, the delay value can be set on both the SRT source device and the SRT target device. The final SRT transmission delay will be the larger one of the two values. Default is 125ms;

Advanced settings: Open and Close;

Encryption mode: AES-128, AES-192, AES-256;

AES key: encryption AES key can be 10-32 letters or numbers combination;

Bandwidth overhead: It is set as the percentage value based on network link quality. Using this value to multiply the total bitrate of the audio and video encoded by the encoder, this will get the occupied maximum bandwidth allowed by Bandwidth Overhead. This value plus the total of video and audio bitrate is the threshold of the current SRT transmission bandwidth, and also the maximum bandwidth that SRT stream can be used. From the perspective of "overhead", it is the extra "invalid" bandwidth to be used in addition to the media content required for transmission (which can be understood as the payload), but it is different from our common protocol overhead, TCP header overhead, UDP header overhead. The bandwidth overhead here is not a fixed 20~60 bytes TCP header overhead or 8 bytes UDP header overhead. It changes in real time according to the network conditions. The worse the network link conditions are, the more overhead needs for normal transmission. The range is 5%~100%, and the default overhead is 25%;

Load size: Sending data packet size, it is optimal that the receiving port should equip with the same size. The default size is 1316, the optimal packet for the codec.

Note: Below format is the reference delay value under different packet loss. When $RTT > 20ms$, it requires to increase the delay appropriately.

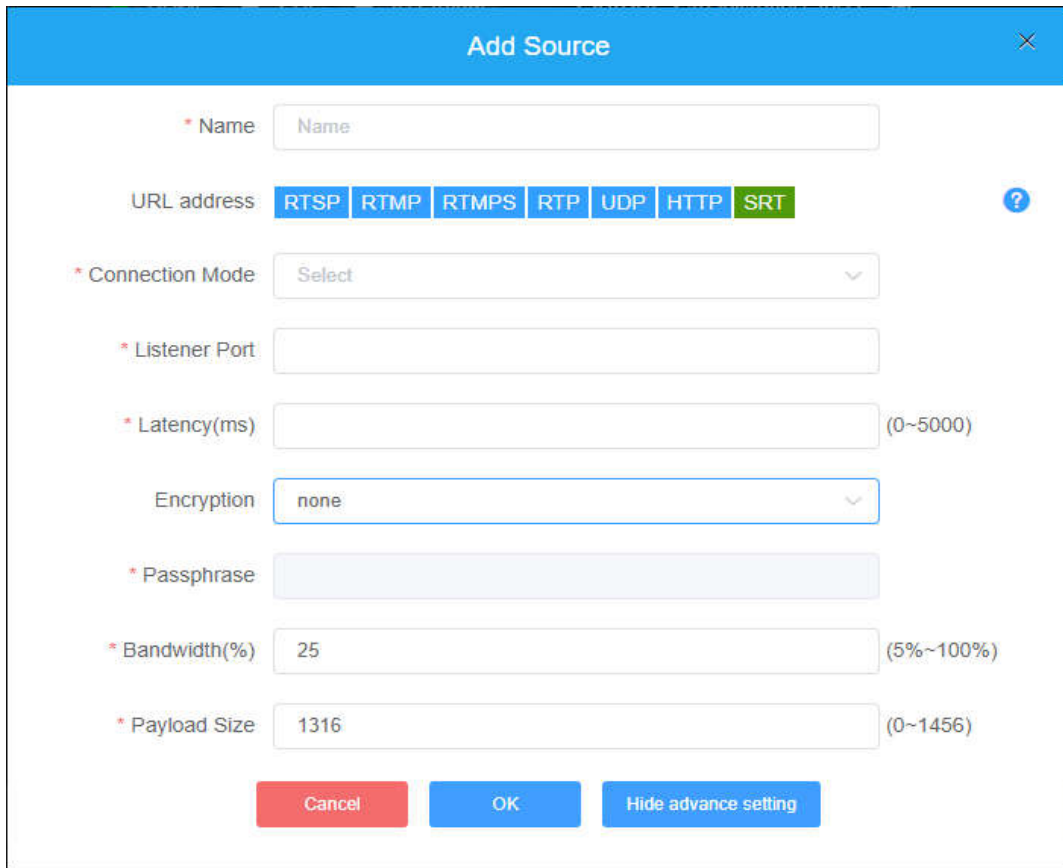
Network highest packet loss rate (%)	RTT Multiplier	BW Overhead	The lowest SRT delay (when $RTT \leq 20ms$)
≤ 1	3	33	60
≤ 3	4	25	80
≤ 7	5	20	100
≤ 10	6	17	120

3.4 Save the settings, then a SRT-TS pushing address will be added. The encoder starts to initiate a handshake connection to the receiving port.



4. Decoder SRT function configuration

4.1 Add a SRT source in the page of video decoder by “Media”--“Add Source”--“SRT”.



Name: You can name it freely, supporting letters and numbers combination;

Connection Mode: Listener;

Port: Configure a listening port which should be consistent with the sending port;

Latency: please set it according to the configuration of the video encoder and the final delay will be the larger one between encoder and decoder.

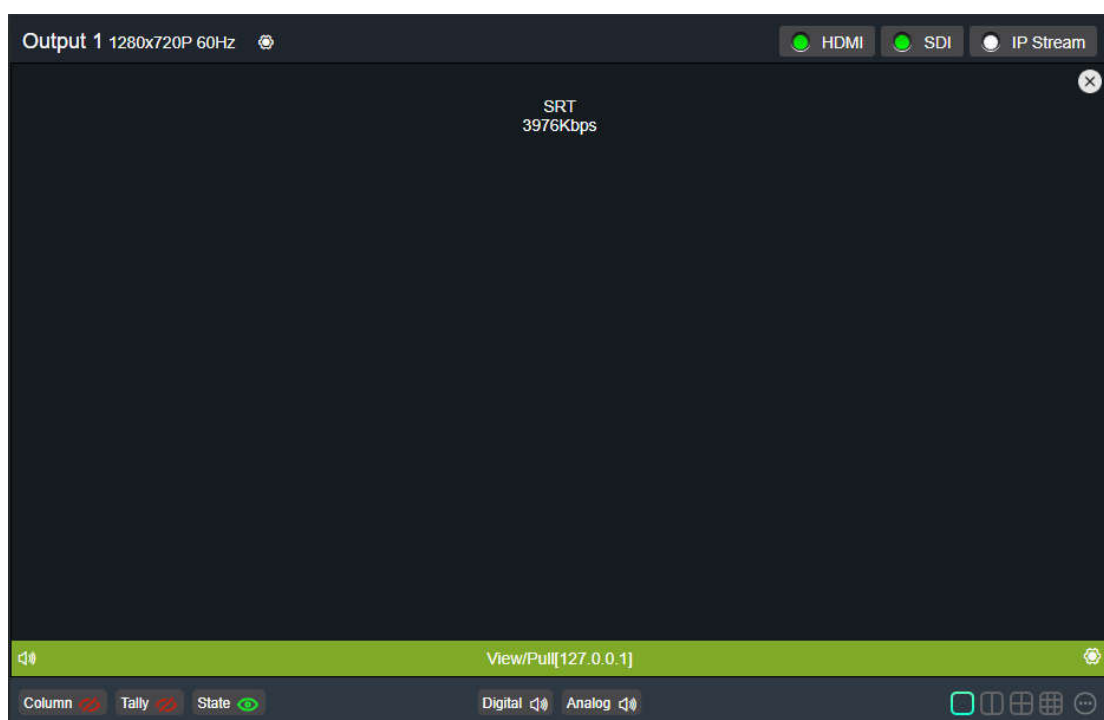
Default is 125ms;

Encryption Mode/AES KEY: AES-128, AES-192 and AES-256. It should be the same as the sender;

Bandwidth Overhead: Please set it according to encoder, the default one is 25%;

Payload size: Please keep the same settings as the sender, the default one is 1316.

4.2 Please add the SRT source to the output window, and the decoder starts to listen for SRT requests, it will establish the SRT connection while receiving the right SRT “Call”, and decode then output video.

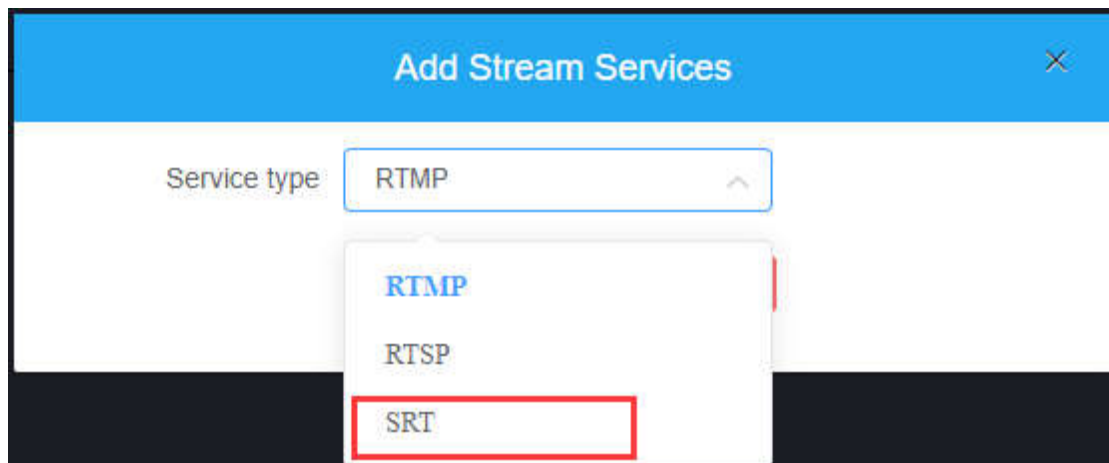


5. Media Gateway SRT protocol Conversion

Media Gateway (such as Kiloview D300Plus/Pro) can realize the video stream protocols conversion between SRT, RTMP, RTSP and some other protocols.

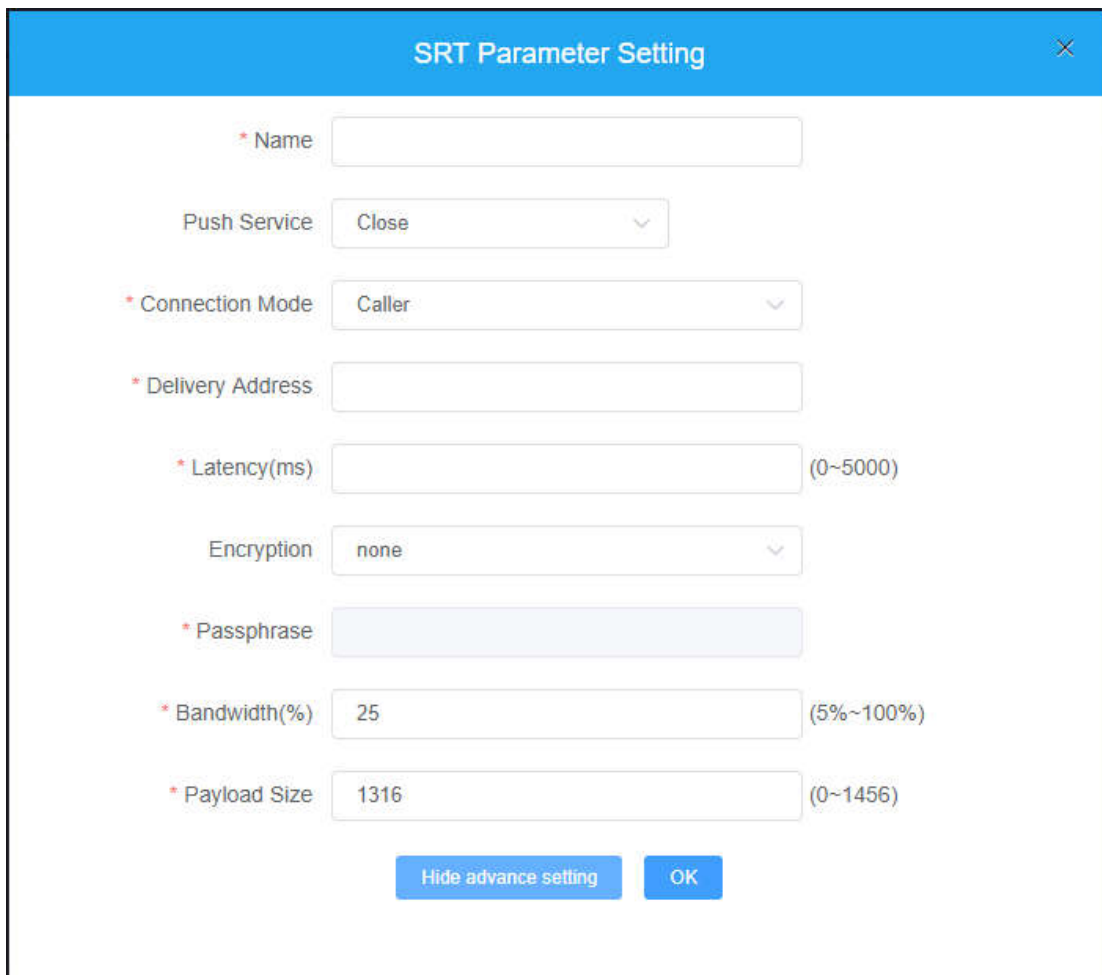
5.1 If RTSP/TS-UDP/RTMP/HLS/SRT to SRT, that is, the decoder works as the “Caller” of SRT, and connect with other SRT receivers.

1) Add a stream service in the publishing points and choose SRT, click “OK”.

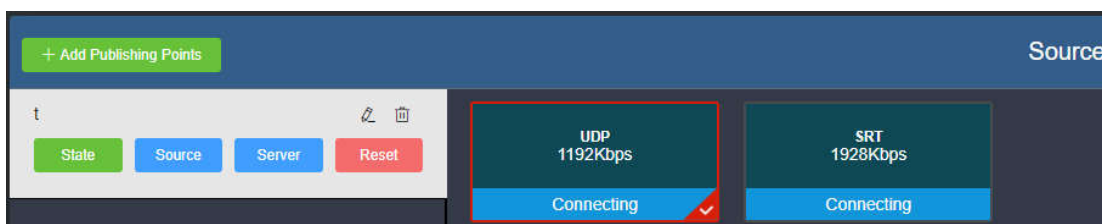


2) Enter the SRT configuration page, the SRT parameters can refer to that of the encoder.

Notes: Delivery address format: srt://IP address: port

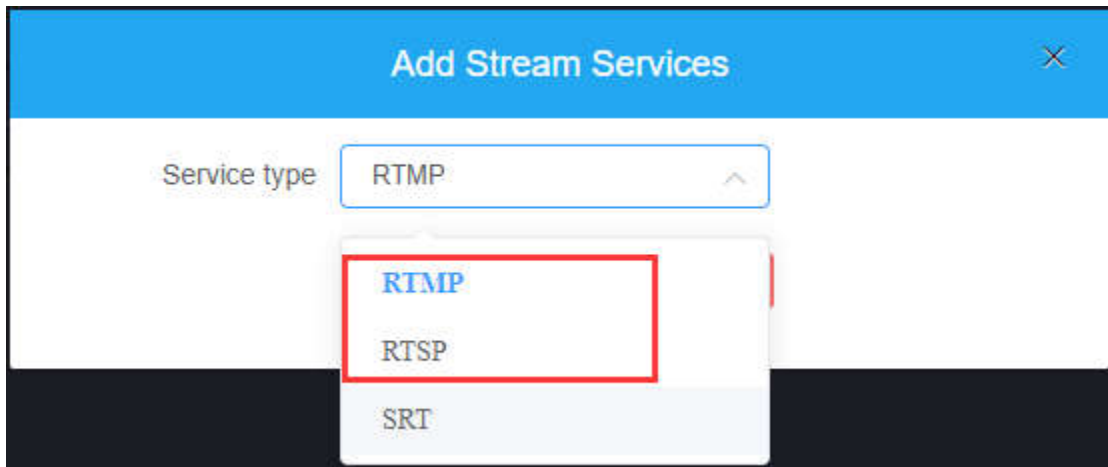


3) In “Choose a source”, you can choose the stream that you want to convert from the source, then the stream can be converted to SRT stream.

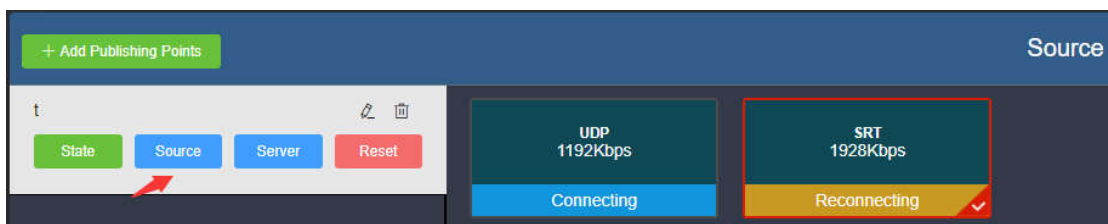


5.2 If SRT to RTSP/RTMP, it will convert or distribute the SRT received in local, and then convert it to RTSP or RTMP pushing.

1) In “Add Stream Services”, if adding type is RTMP or RTSP, it can convert SRT to RTSP/RTMP.



2) In “Choose a source”, please choose the SRT stream that you want to convert in the source, then it will be converted to RTSP/RTMP stream.



Thank you for reading.

Changsha KILOVIEW Electronics CO.,LTD.

Phone: 86-731-88315979

Website: en.kiloview.com

Support email: support@kiloview.com

WhatsApp: +86-18573195156/18573195256

Skype: support_72698/keqiu wang

Address: 188#, Middle Huanbao Rd, Yuhua District, Changsha, Hunan Province, China