BLANKOM® an IRENIS brand ...setting signals

SDE-(1)265

SoC Encoder & IPTV Streamer with HD-SDI Input

h.265 and h.264 compatible encoder & IP streamer as boxed or 19'' 1RU

- HD-SDI or SD-SDI input for encoding
- Stereo Audio embedded or external Input (3.5mm stereo)
- HD Resolution 1080p, 1080i, 720p
- IP output: RTSP, RTMPs, UDP/RTP, HTTP, HLS, FLV, SRT, MJPG
- Distribution of Video Camera HD-SDI and other sources content over LAN, WAN or internet.
- 2 or max. 4 simultaneous and independent Live stream broadcast encoder engines to multiple destinations
- Video-over IP applications (Studio signal distribution)
- IPTV/OTT applications
- Video conferencing, Camera streaming
- IPTV on LAN applications, Corporate IPTV for Broadcasters
- HD and SD video encoding (incl. 1080p)

Complementary products:

- HSD-340 HDMI to SDI converter
- HDD-275 Decoder IP to 4K/HDMI/HD-SDI/VGA/CVBS outputs
- IPTV Set Top Box 6800+
- HDC-5004: IP UDP/RTP to 4 muxed adj. RF DVB-C Modulator
- HDC-5016: 512 IP to 16 DVB-C channels

BLANKOM SDE-(1)265 encoder serves the
distribution of SD and HD TV/video content
through IP networks in digital quality.
The live video can be received
by Internet media server
by TV sets with IPTV Set-Top Boxes,
on PC has and tablets with VLC Player.

BLANKOM SDE-265 & SDE-1265

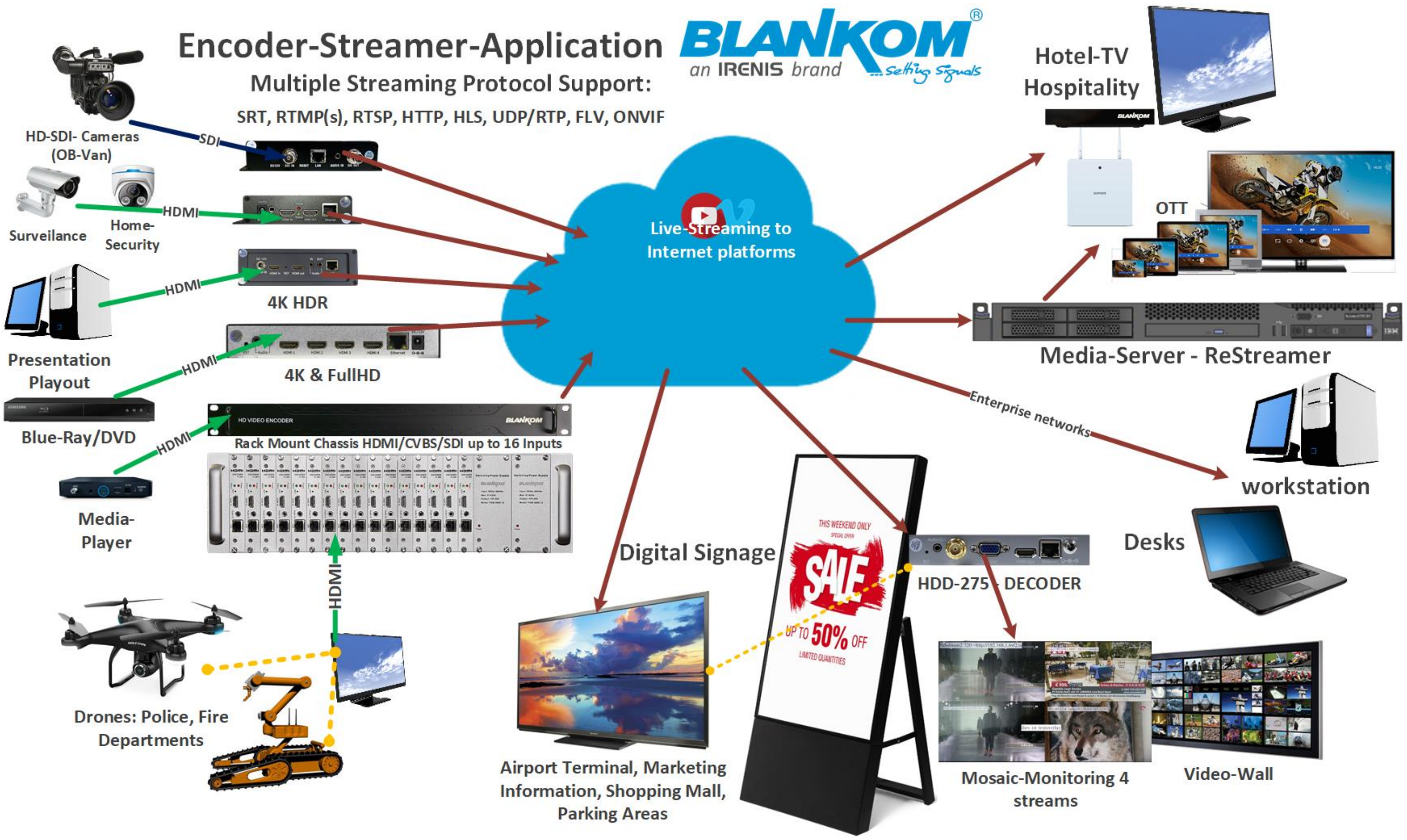
IPTV encoder is designed for TV signal distribution in excellent quality over LAN and INTERNET.

The h.265 (HEVC) compatible compression technology features low-latency and low bit rates for IPTV/OTT systems. The high-efficient encoding chips saves bandwidth cost through all its resolution range.

Distribution of SD and HD TV channels through the IPTV/OTT network using state-of-art IP technology from almost any kind of video input.

Excellent video and audio quality. High reliability. No regular service and maintenance need during operation.
Remark: SDE-264 is not available any more

info@blankom.de www.blankom.de



Function	h.265 (HEVC compatible) and h.264 (MPEG4 compatible) Encoder and IP Streamer
INPUT	3G HD-SDI/SD-SDI (BNC type) input and loop through output, SDI level A+B autodetect*
Resolution	1080p, 1080i, 720p and below
Video encoder	h.265 (HEVC) or h.264 (AVC) or MJPEG compatible
Audio encoder	AAC, AAC++, MP3, MPEG1Layer2, AC-3 stereo compatible
Audio Bit-rate:	Bit-rate: 32k/48k/96k/128k/160k/192k, Data-rate: 64 kbps-384 kbps
Data interface GbE	RJ45, 1000BaseT Ethernet interface, management by web browser
Protocol	HTTP, RTSP, RTMPs, UDP/RTP, FLV, HLS ; unicast/multicast, SRT P2P
Streaming	DVB-conform IP streams w/ Tables and selectable adding Zero-PID function
Data Rate	32 kbps – 32 Mbps
Encoding bitrate process	CBR or VBR (CBR setting enables zero-packet insertion for the TS)
SMPTE 425	Support Level A & B
GOP Structure	IBBP
ONVIF 2.x	Supported by RTSP: G711A
Picture adjust	De-interlacing, Noise reduction, Sharpening
OSD	4 Logo and Text Insertion as transparent overlays possible
Power supply	12V DC, 1A
Dimensions	165x85x24mm
Weight	0.5 kg
Consumption	5...10W

Anmerkung:

Alle von uns veröffentlichten Betriebsanleitungen richten sich an den Antennen- und IT-Fachmann, der über grundlegende Kenntnisse der Empfangs-, Netzwerk- und Anlagentechnik verfügt. Die Einhaltung aller relevanten Vorschriften und Richtlinien für den Aufbau und Betrieb von solchen Anlagen obliegt dem Installateur und/oder dem Betreiber. Insbesondere sind die in den jeweiligen Ländern geltenden Vorschriften und Richtlinien für die Inbetriebnahme speziell für den Stromanschluß und alle mit den Produkten in Zusammenhang stehenden und geltenden Normen und Gesetze einzuhalten.

Remark:

All operating instructions published by us are intended for the antenna and IT specialist who has basic knowledge of reception, network and system technology. Compliance with all relevant regulations and guidelines for the installation and operation of such systems is the responsibility of the installer and/or the operator. In particular, the regulations and guidelines applicable in the respective countries for commissioning, especially for the power connection, and all standards and laws related to the products must be complied with.

Annotation:

Tous les modes d'emploi que nous publions sont destinés aux professionnels de l'antenne et de l'informatique qui ont des connaissances de base en matière de réception, de mise en réseau et de technologie des équipements. Le respect de toutes les réglementations et directives pertinentes pour l'installation et l'exploitation de ces systèmes relève de la responsabilité de l'installateur et/ou de l'exploitant. En particulier, il convient de respecter les réglementations et directives applicables dans les pays respectifs pour la mise en service, notamment pour le raccordement électrique, ainsi que toutes les normes et lois relatives aux produits.

Annotazione:

Tutte le istruzioni per l'uso da noi pubblicate sono destinate al professionista dell'antenna e dell'informatica che ha una conoscenza di base della tecnologia di ricezione, di rete e delle apparecchiature. Il rispetto di tutti i regolamenti e le linee guida pertinenti per l'installazione e il funzionamento di tali sistemi è responsabilità dell'installatore e/o dell'operatore. In particolare, devono essere rispettati i regolamenti e le linee guida applicabili nei rispettivi paesi per la messa in funzione, soprattutto per il collegamento alla rete elettrica e tutte le norme e le leggi relative ai prodotti.

Anotación:

Todas las instrucciones de uso publicadas por nosotros se dirigen al profesional de la antena y de la informática que tiene conocimientos básicos de recepción, de redes y de tecnología de equipos. El cumplimiento de todos los reglamentos y directrices pertinentes para la instalación y el funcionamiento de dichos sistemas es responsabilidad del instalador y/o del operador. En particular, deben cumplirse los reglamentos y directrices aplicables en los respectivos países para la puesta en marcha, especialmente para la conexión de la energía y todas las normas y leyes relacionadas con los productos.

Anotação:

Todas as instruções de operação publicadas por nós são destinadas ao profissional de antena e TI que possui conhecimentos básicos de recepção, rede e tecnologia de equipamentos. O cumprimento de todos os regulamentos e diretrizes relevantes para a instalação e operação de tais sistemas é de responsabilidade do instalador e/ou do operador. Em particular, os regulamentos e diretrizes aplicáveis nos respectivos países para comissionamento, especialmente para a conexão de energia e todas as normas e leis relacionadas aos produtos devem ser obedecidas.

View the new add-ons and features: They are written as add-on pages to the last manual-page (just before the addendums)

**) example: SDI-Auto-detection of Level -A or -B implemented since Firmware Version 5.17*

Quickstart   Variant with SDI Inputs: SDE-265 (no 264 version available any more)

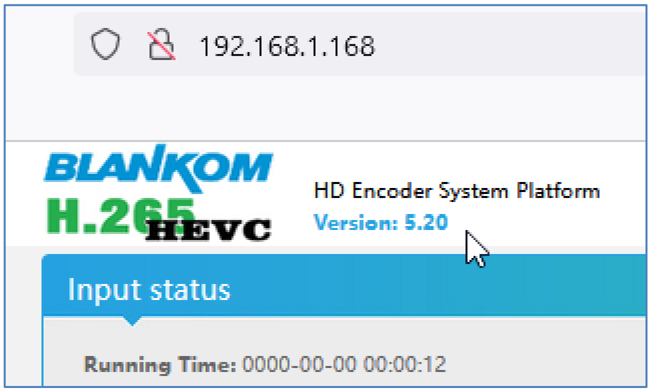
Hardware and Software versions from the past and until now:

The old and new (temporary) design varies a little:

Produced until November 2020:

The Firmware Version ends with 6.51E (HDMI) and 6.51S (SDI Input). We were forced to redesign the hardware SoC's because of not available chipsets (Worldwide Chip-situation). We were forced to source from 2 different PCB and housing suppliers because of being able to serve our customers in time avoiding delivery times of 3 month and more…

The next Version in Black housing -but same lookalike- got Firmware versions with 5.xx and ended with 5.20 for HDE-265L and SDE-265 as a common firmware in both models: Can be detected on Status-page when entering



which also is the same Version in this temporary hardware design:

Temporary just until August 2022, FW 5.20

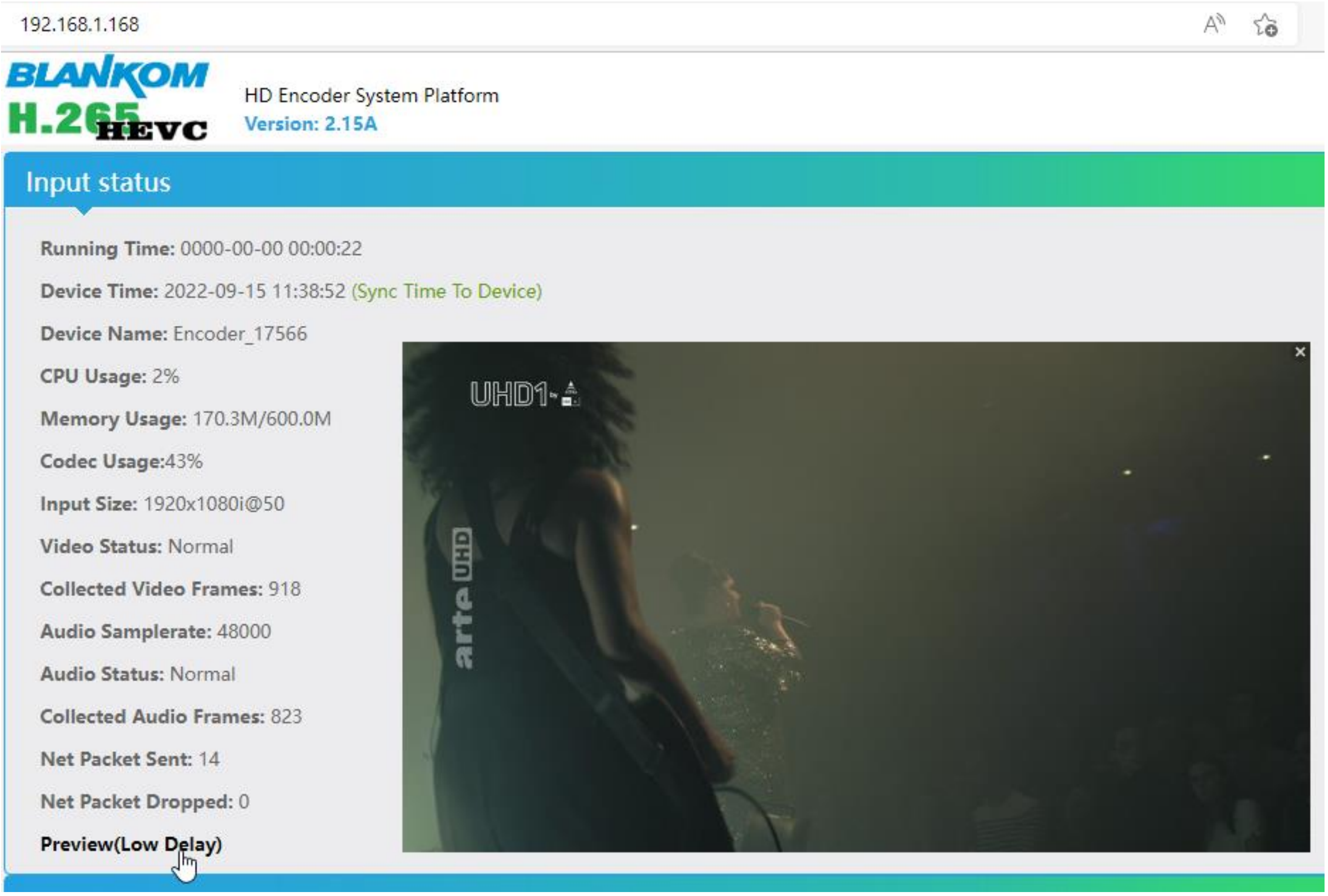
The new Firmware Versions HDMI=SDI has started with 2.** and begun from 2.13A; where 'A' indicates the new modern chipset from Ambarella (SDI HDMI VGA - same firmware) which is available in 2 different housings.

Actual

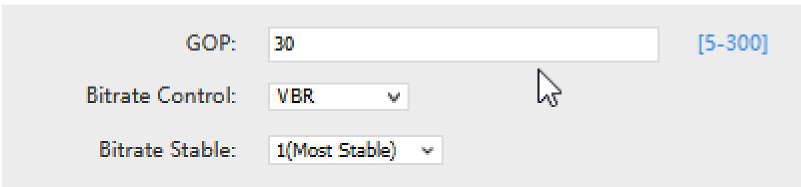
Design

Obvious: The 3 dimensions of the variants are all different…

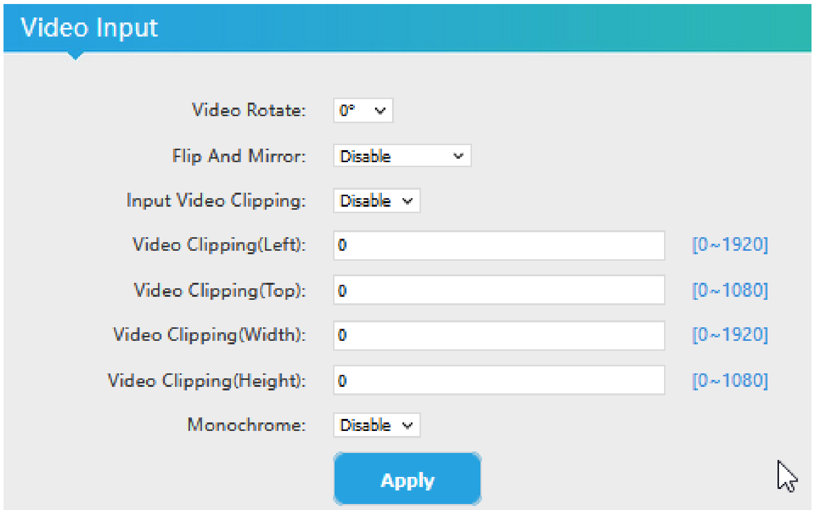
The new Chipset has some advantages and improvements like a direct preview Window as WebRTC:

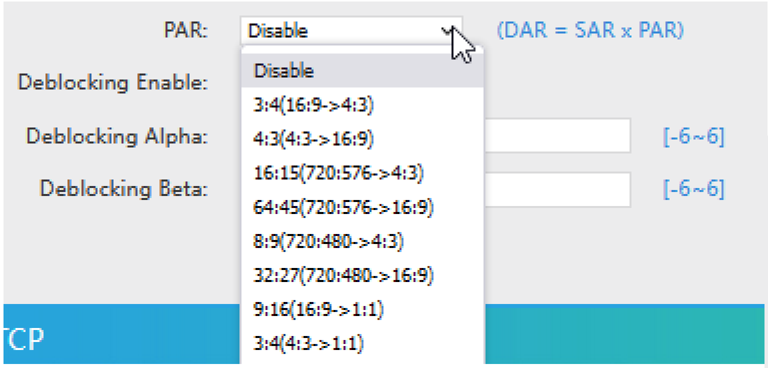


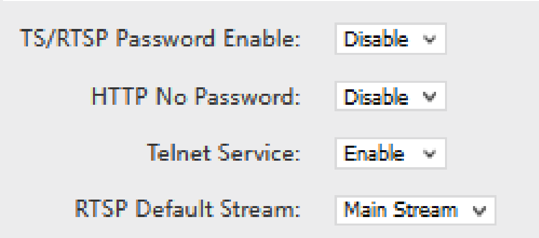
GOP settings:



Video changing selections:







All SDI-Versions have a loop through to cascade the Input to other SDI-devices... do not mess up with them...

ATTENTION:

Please do not feed the SDI BNC Input with remote powering like some Cameras doing with 12V DC. This will destroy the SDI Input circuit. In addition, before connecting the 12V DC Power source, connect the Ethernet and SDI-BNC cables first. A hot plug of SDI BNC is not recommended because of non-grounded DC potential differences (over a long coax cable) might destroy the input circuit.

Sticker with default settings (MAC may be different)

Notes and Hints:

The Gigabit-Ethernet-port might not support PoE so please take care of not accidently using a PoE switch- you can damage the port and the unit will be not accessible anymore.

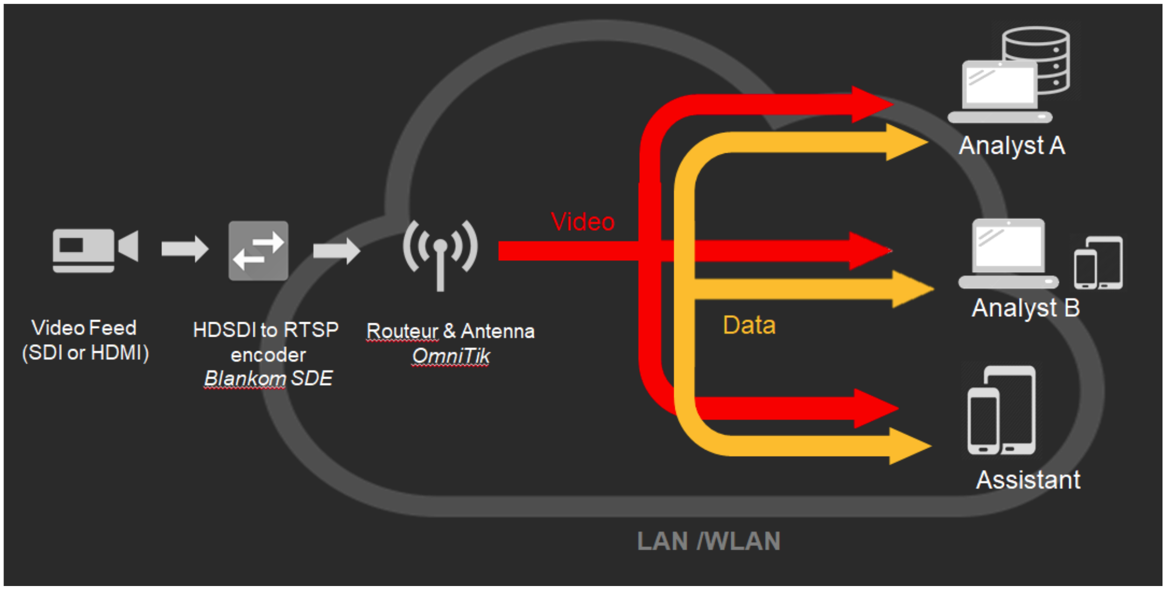
We recommend using an IGMP-V2/3 protocol, capable GBE- Switch to avoid flooding your network with unmanaged multicast streams. In addition, some consumer Internet routers do not like Multicasts (UDP/RTP) and might reboot periodically.

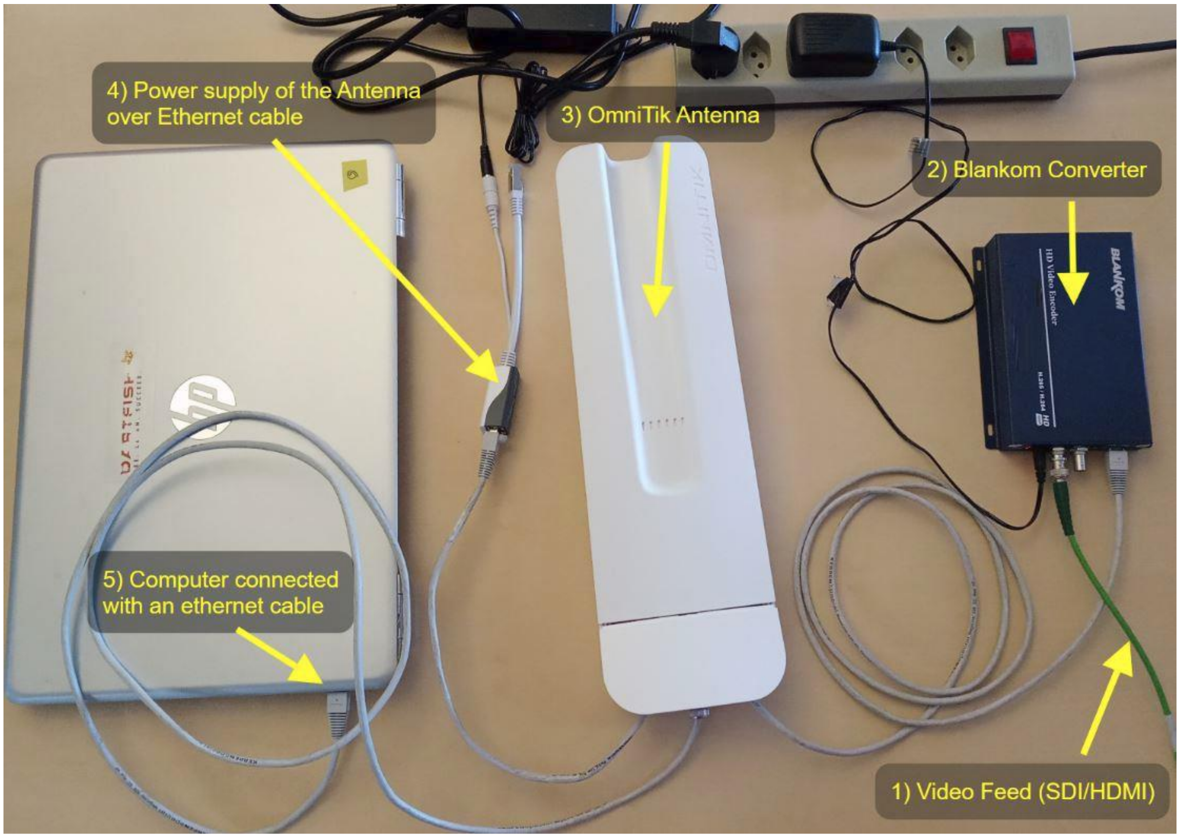
An Internet-connection is not necessary as long as you need to use NTP and does not have an own NTP server in your network.

Please assure that your HDMI –Output you like to encode is set to max. HD with 1080p60 or lower. Higher values will not work.

The embedded Linux system takes some seconds to fully boot. After the System-LED is on, you can connect your browser to it. We recommend Chrome, Opera, and Mozilla. For a preview Popup in the browser, a flash-player add-on need to been installed for the browser.
Sometimes it is helpful to reload the browser – page to get the changed settings and values because of different browser behaviours...
The RESET button will erase all your settings and the unit will been forced to start with factory defaults. Use a thin wire to pass the small hole and press the inside button by it for at least 5-10 seconds until the System LED will go off. The encoder would perform a restart than after releasing the button.
The Web-Interface lookalike may vary between different Versions but it is self-explaining.
The SDI versions supporting the first embedded Stereo-Audio-Pair to be encoded.
PLEASE check grounding voltage potential differences between the grounding of SDI-Coax, Network Cable and Power-Supply-Ground at the metal chassis. If necessary, you should ground all.

Example with the Dartfish –Setup:

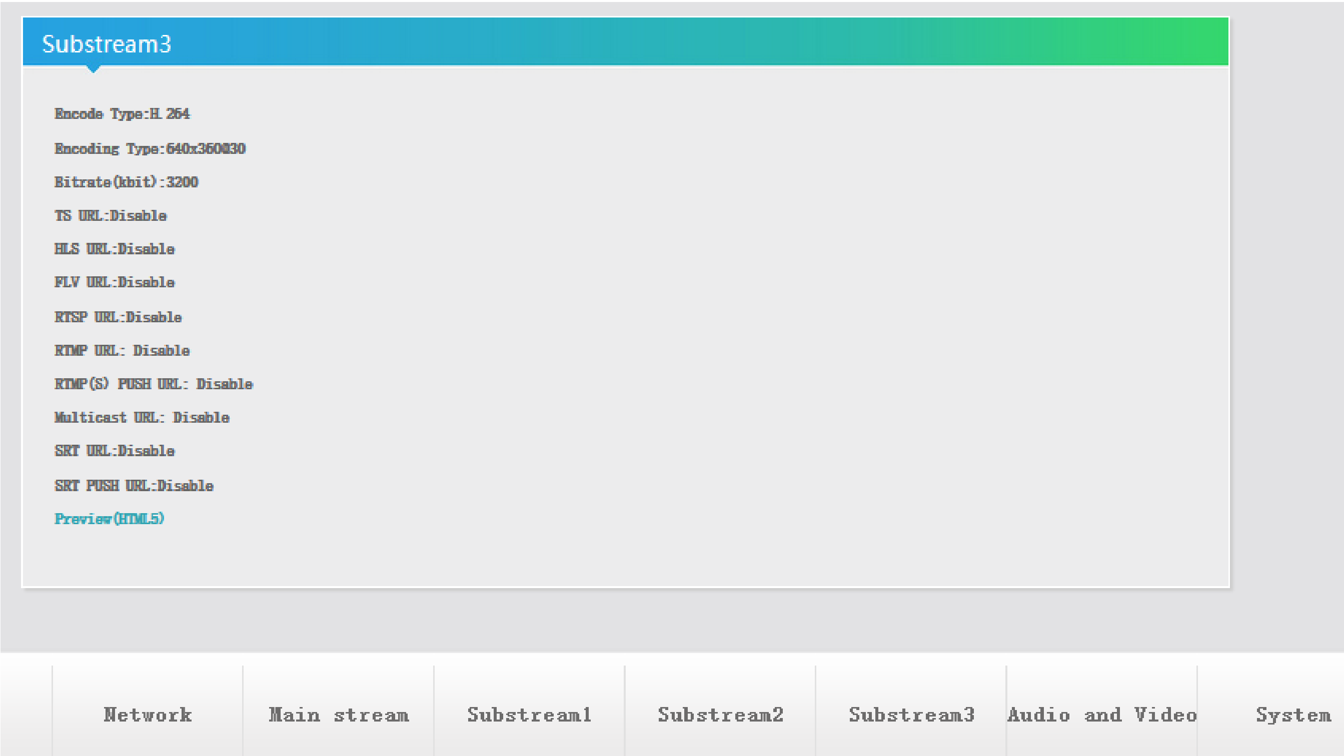




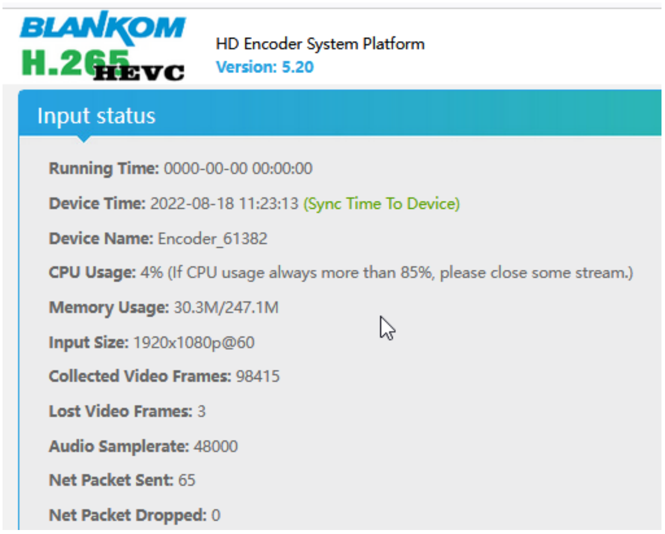
Please assure that the Ethernet port of the HDE-/SDE265 Encoder (Converter) does not get the Power over Ethernet (PoE) from the OmniTik Antenna. We recommend to first configuring the Antenna ports and avoid PoE forwarding on all other ports. Than connect the BLANKOM SDI-Streamer - Converter with all connectors before powering on the whole system.

As mentioned above, we have done 2 *Hardware improvements* of the SDI-Encoder-Version so it comes with a new Firmware as well. (Same for HDE-265, the new is named HDE-265L).
Differences:

- Old Hardware & firmware ends with ***V6.56S* produced until** February/April 2021
- Version 5.xx Hardware with 4 encoder sub-parts looks slightly different:



- Now more encoding parts 1x Main and 3x Secondary are active – but depend from each
- **Next Firmware starts at V.5.xy or actually is:**
  5.20:

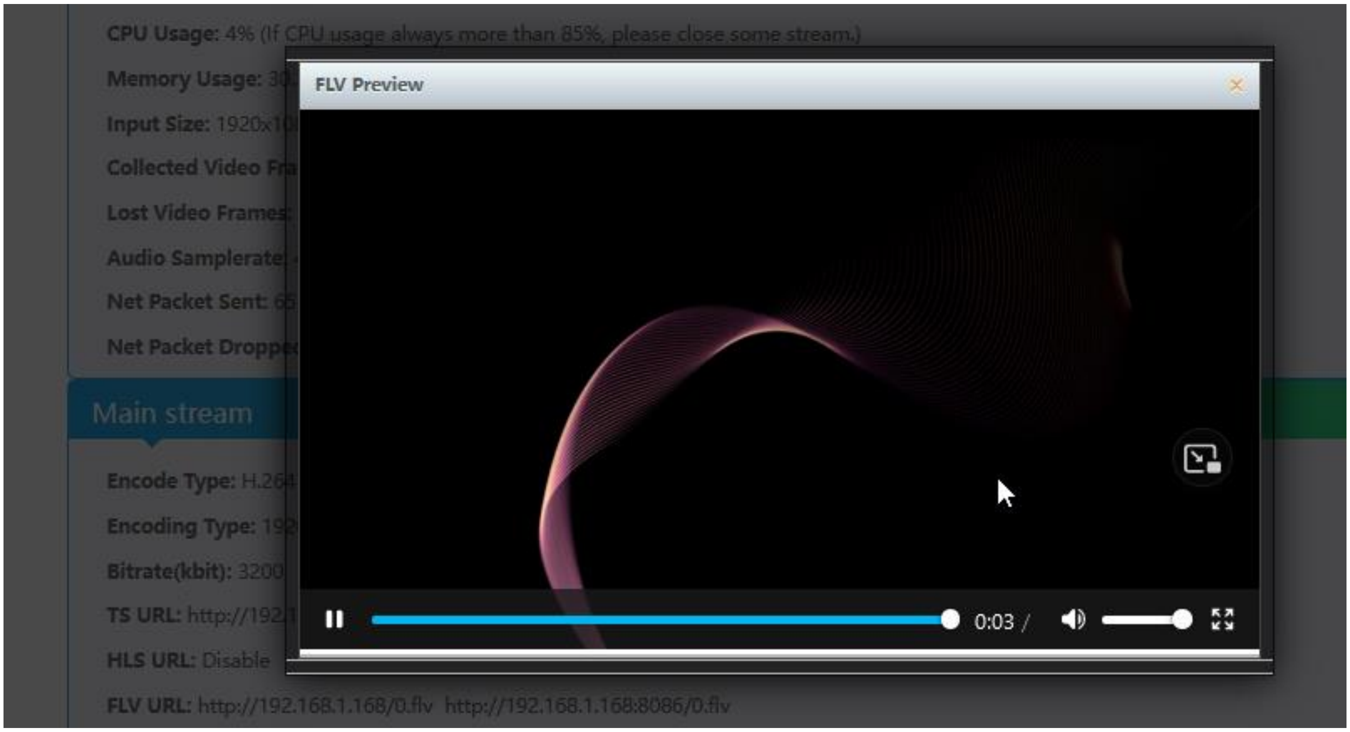


- **From Version 5.11, the web-player preview can be chosen also for h.265 encodings:**

FONTs

Improved for better lookalike and visibility avoiding serifs

even with HEVC/h.265



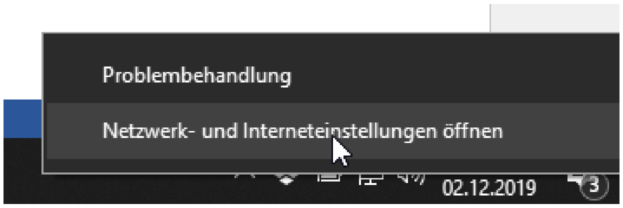
But no Play/stop/ffwd/rwd available Like in h.264 encoding mode:

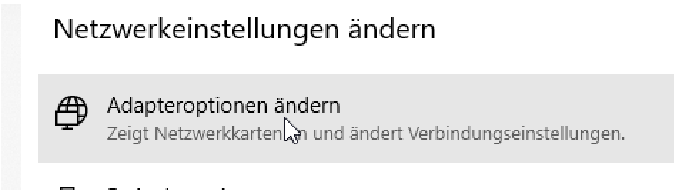
- **A hint:** Sometimes it helps to reboot the unit in the SYSTEM Menu when you have changed essential streaming values like UDP -> RTP… But usually, no need

Setting up your PC/Laptop before connecting:

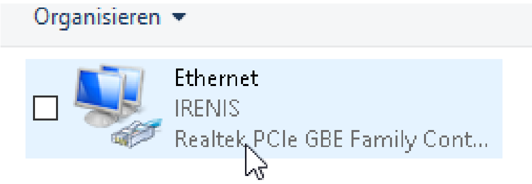
If you use a Windows based PC, you should assign its Ethernet adapter into the same range like the encoder: Use a static IP like follows:

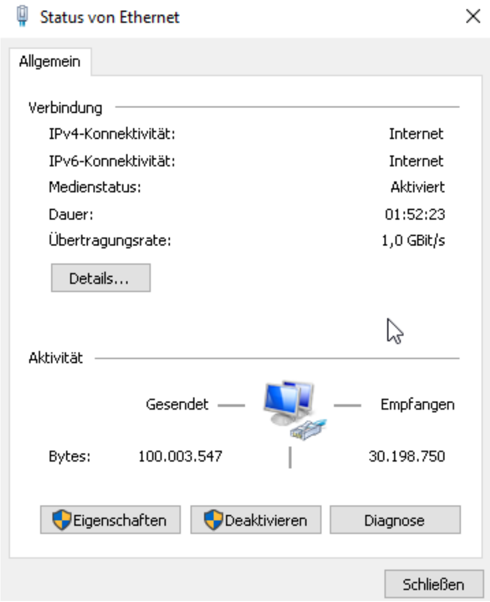
1st: Open your network settings in System Menu:

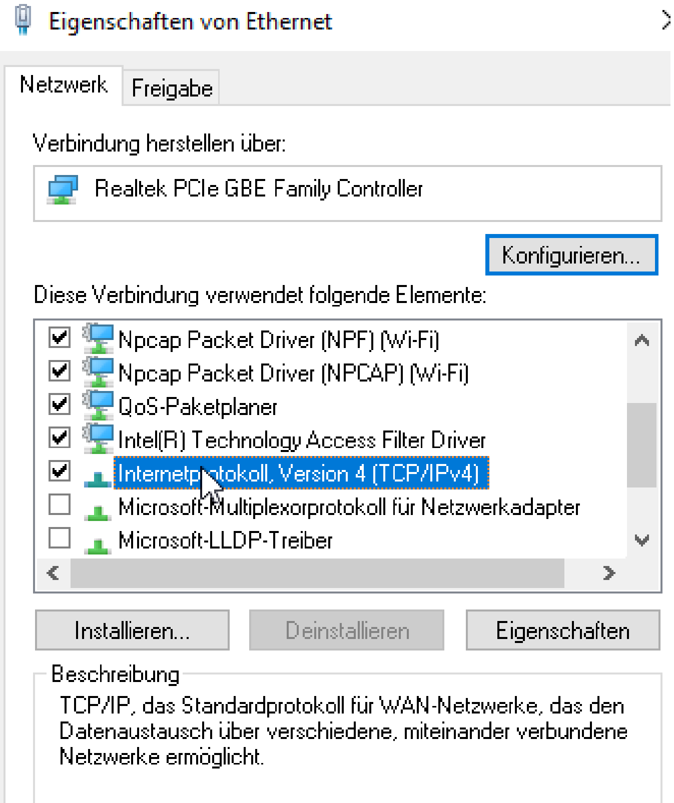




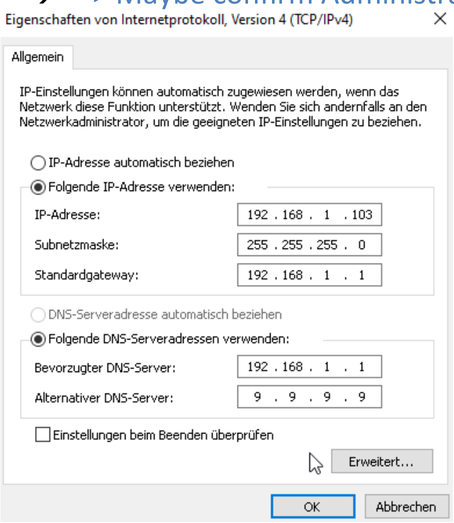
→







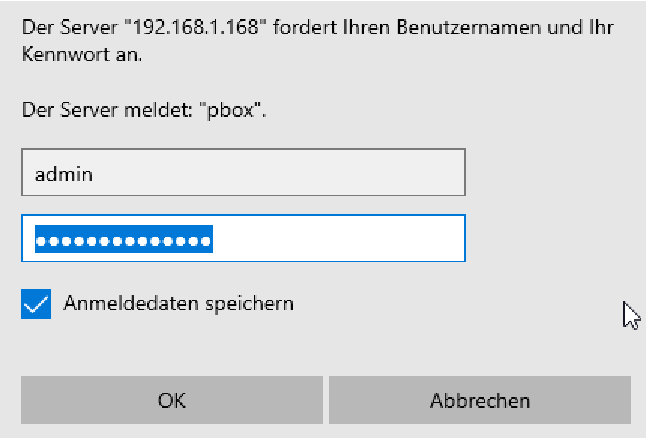
→ -> Maybe confirm Administrator access-> Change IPv4 settings:

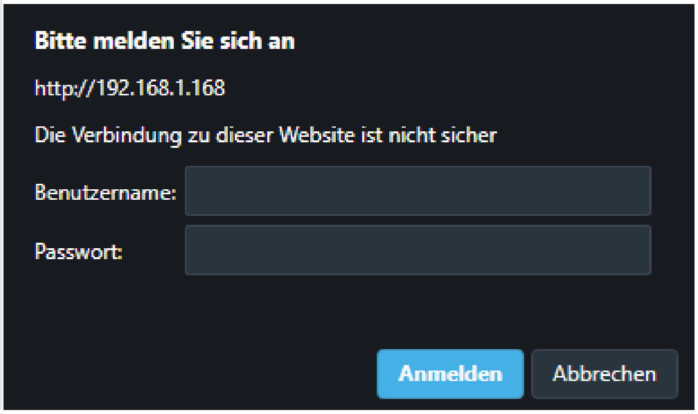


And confirm please. Linux users should know how to change the ethernet or WIFI settings.

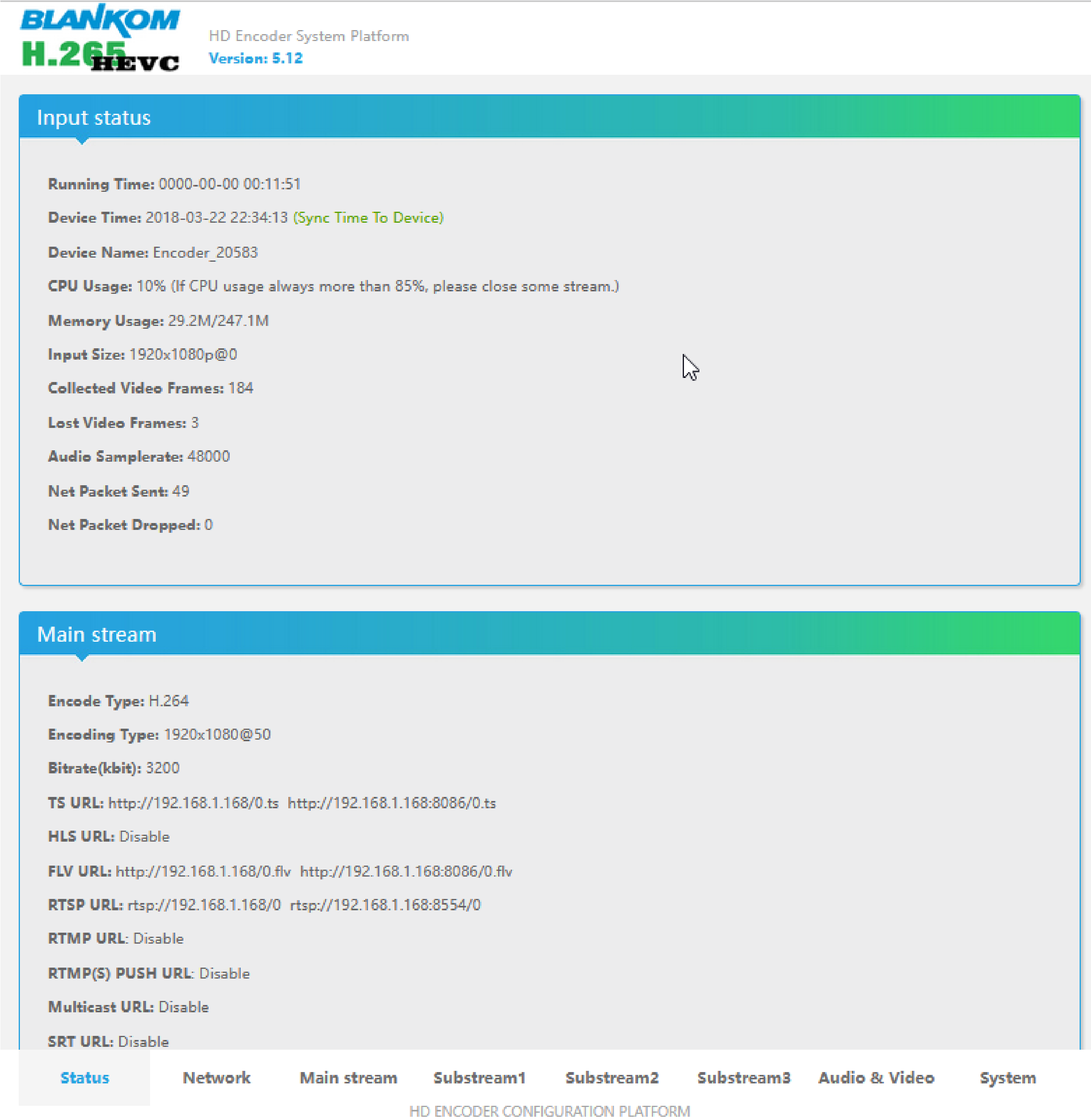
Than open your browser and enter the http- Address of the box 192.168.1.168 (w/o https)

Depending on browser, you will get a log-in-screen window:





Enter the default username = **admin**, default password = **admin** .... and here we go:



If you loose the IP address of this unit you can reset it to default by pressing the RESET button with a kind of needle through the hole in the housing for several seconds until the LED(s) at the netwiork port and or the other status leds will turn partly to off. See explanation above.

Main stream

Encode Type: H.264

Encoding Type: 1920x1080@30

Bitrate(kbit): 3200

TS URL: http://192.168.1.168/0.ts http://192.168.1.168:8086/0.ts

HLS URL: Disable

FLV URL: http://192.168.1.168/0.flv http://192.168.1.168:8086/0.flv

RTSP URL: rtsp://192.168.1.168/0 rtsp://192.168.1.168:8554/0

RTMP URL: Disable

RTMP(S) PUSH URL: Disable

Multicast URL: Disable

SRT URL: Disable

SRT PUSH URL: Disable

Preview(HTML5)

Substream1

Encode Type: H.264

Encoding Type: 1280x720@30

Bitrate(kbit): 3200

TS URL: Disable

HLS URL: Disable

FLV URL: http://192.168.1.168/1.flv http://192.168.1.168:8086/1.flv

RTSP URL: Disable

RTMP URL: Disable

RTMP(S) PUSH URL: Disable

Multicast URL: Disable

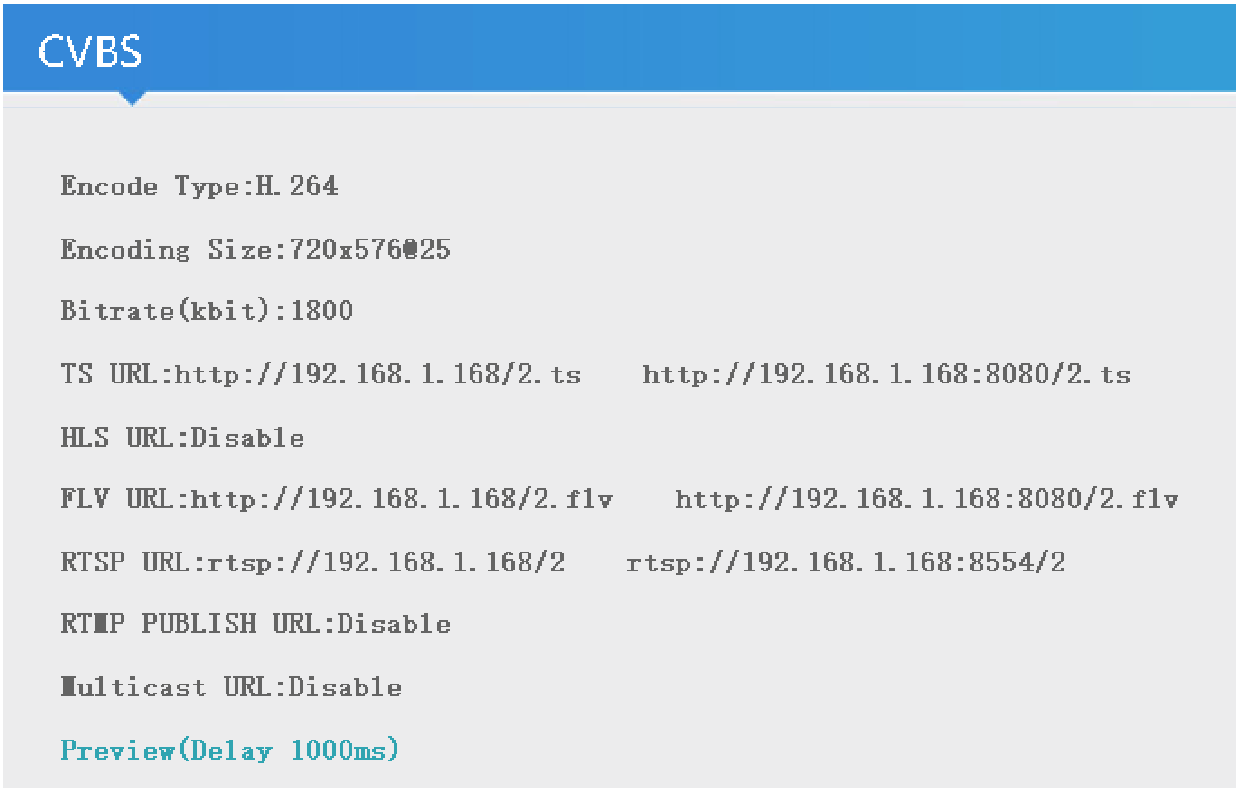
SRT URL: Disable

SRT PUSH URL: Disable

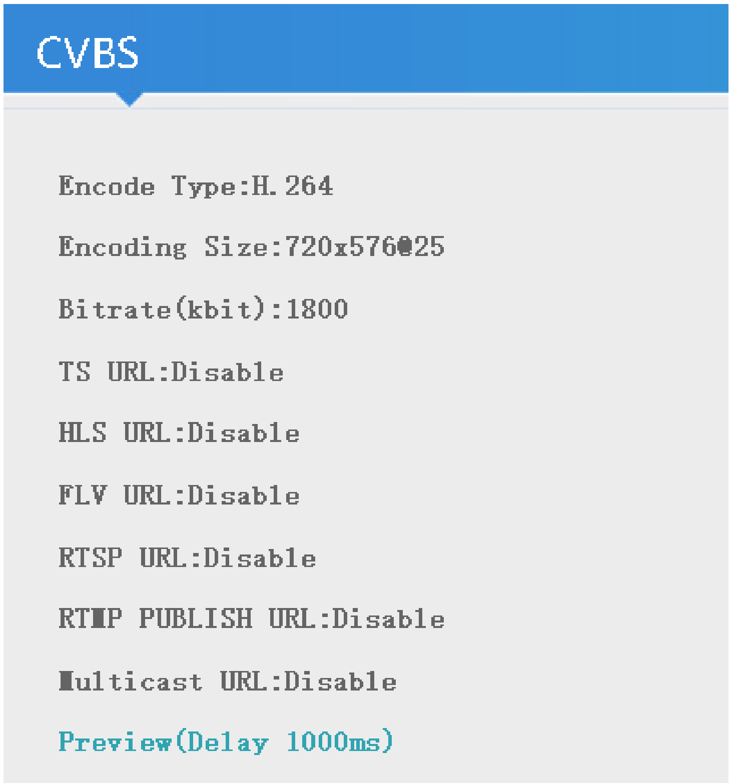
Preview(HTML5)

The Delay depends on codecs/resolutions in use

CVBS settings apply only for ADE-264/265:



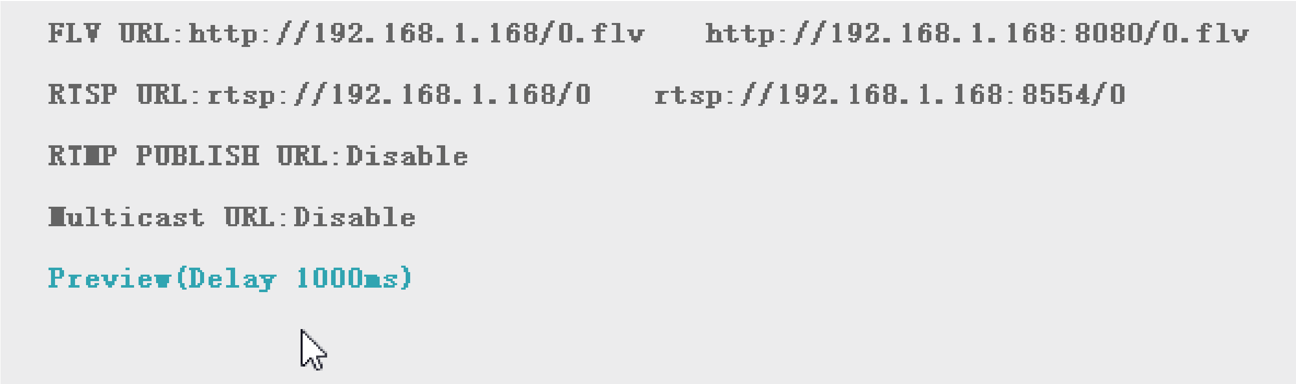
and second:



The STATUS page shows your Setup encodings for the entire MAIN and the Substream(s).

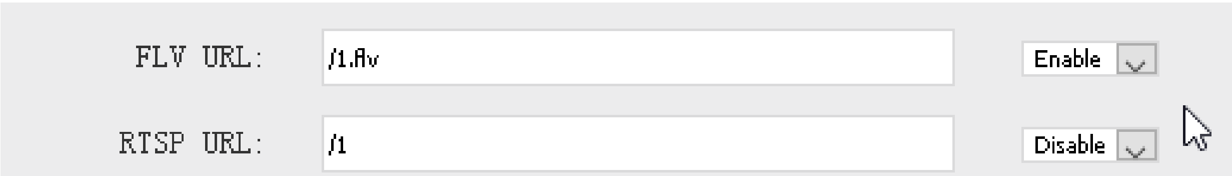
Parallel and different streaming can be used for all encoder parts as long as the capacity of the system is not claiming it: You will get a message if the encoding capacity will be reached and one or more Substream would be disabled... The B-Models support only one streaming Method enabled in Main and sec. Stream (= max. 2 outputs)

In some Sub-Streams Info-sections (model depending) you can check the Picture/Sound directly in the browser by this button:

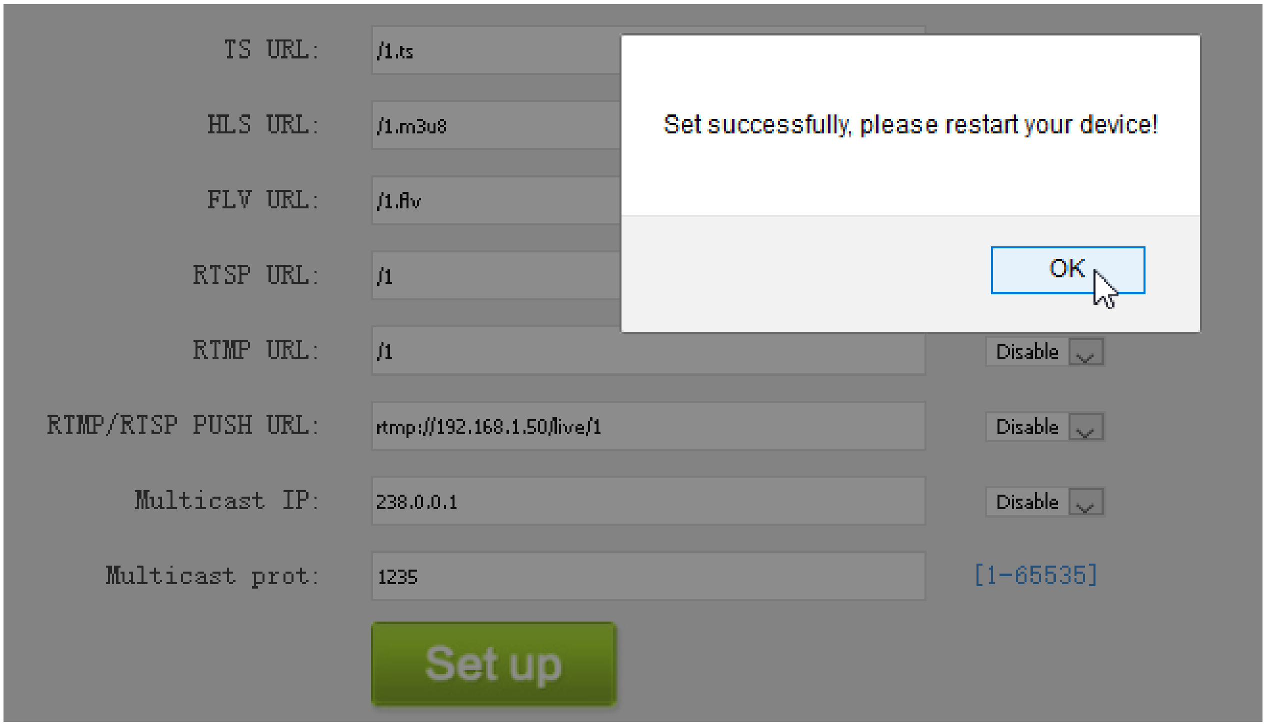


However, you need to enable the FLV or HLS stream before using that – and your browser needs **Flash-Player support:**

Enabling it in the related Sub-Stream settings



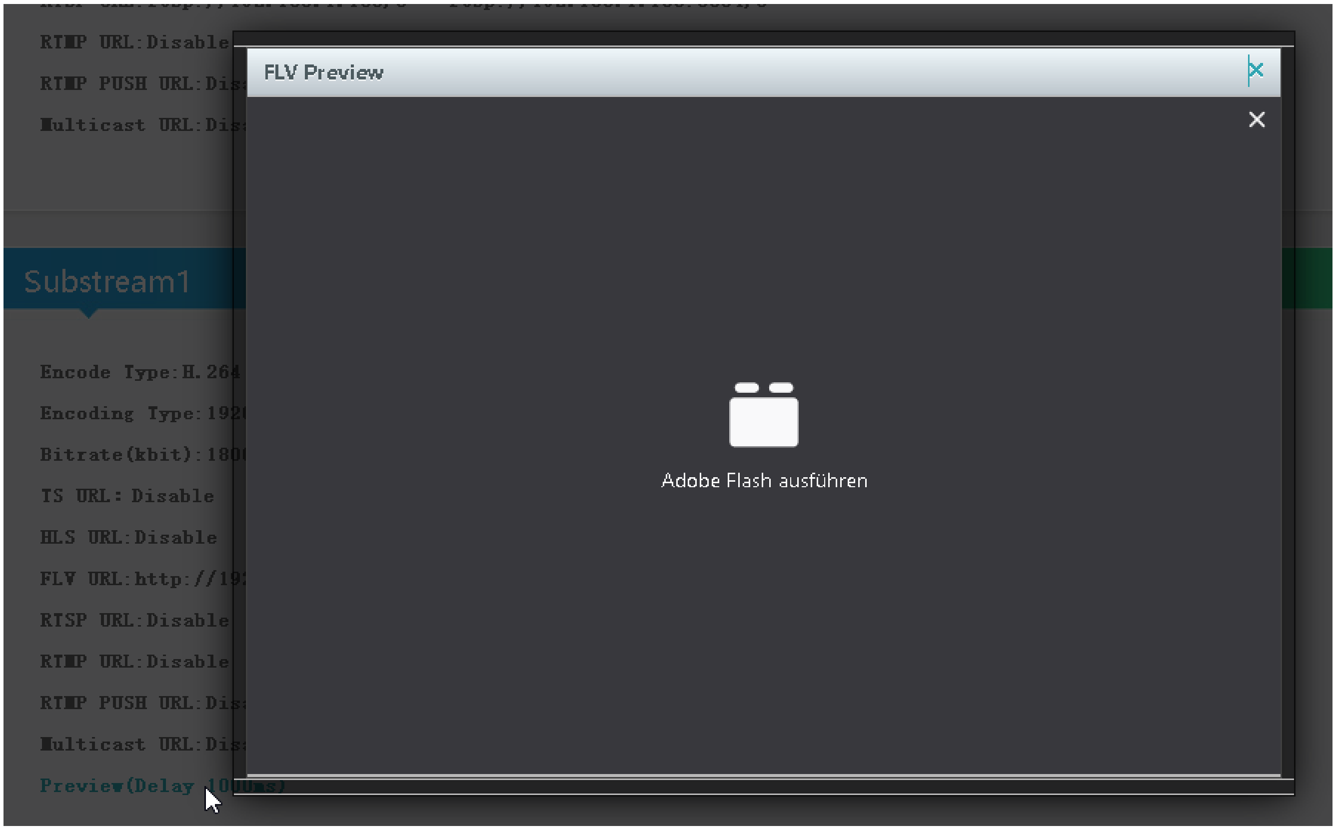
-> Applying it by Set Up!



This does not mean to restart the encoder but to **restart your Stream-receiver-Decoder** like VLC or IPTV Decoder / SetTopBox to re-sync it to the new codec values. This message will pop up every time you change the encoder parameters. Receivers are stupid and might not react to the changed values by themselves.

Depending on Model: Preview in Browser is possible from within the status page as a link:

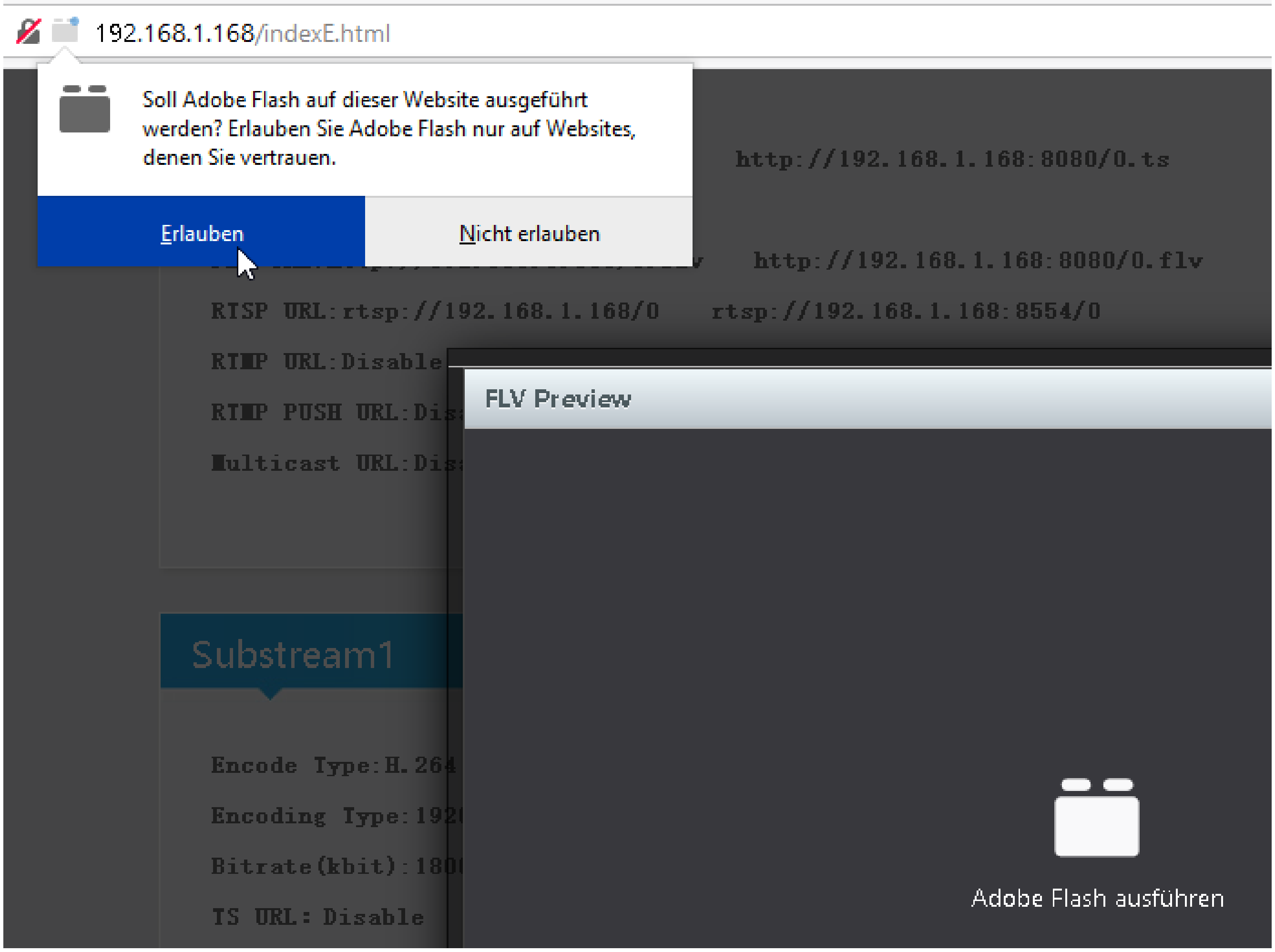
HINT: Adobe Flash **does not work with HEVC** h.265 codec!!!! You need to have h.264 encoding to be set in the main or sub-stream menu (model depending).



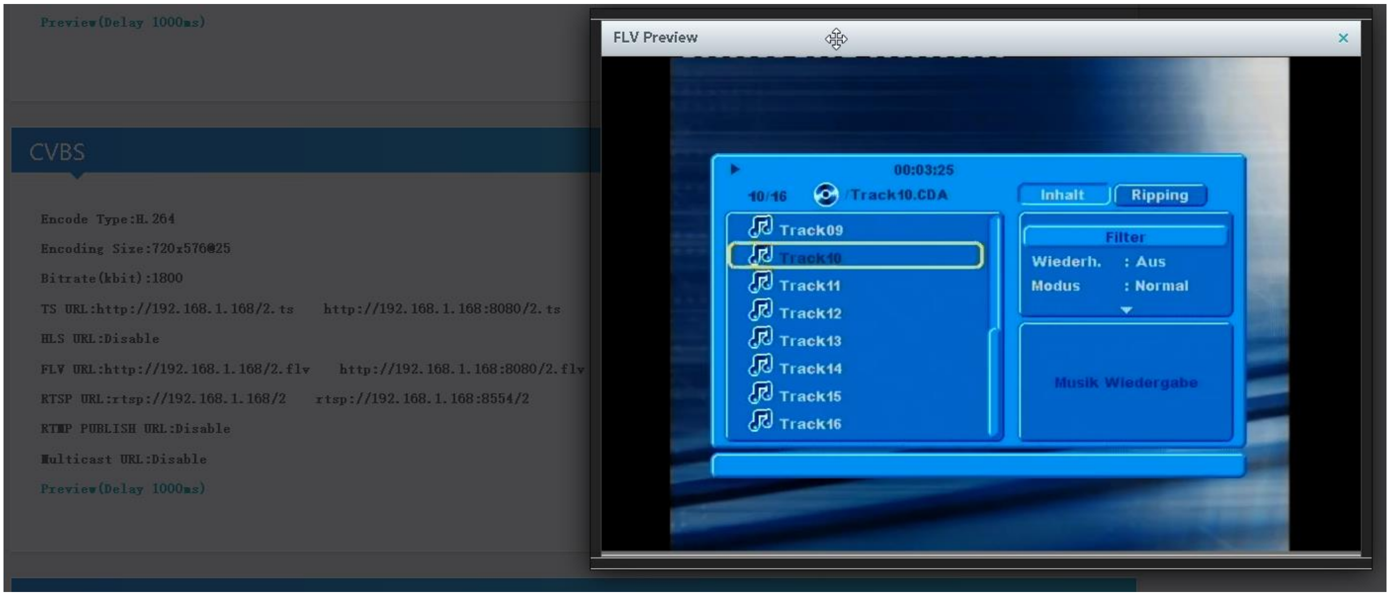
Same happens for HTML5 Preview: h.265/HEVC is not supported but h.264/MPEG4 Part10!!!

Anyway, ADOBE and Microsoft have disabled meanwhile any FLASH as embedded in browsers or similar plugins So we have left only the HTML5 preview.

Allow your browser to do that (here Mozilla):



CVBS-Preview with connected DVD-Player **(ADE-264 only)**



Remark: *Flash-player disabled by **Adobe** since Jan 2021- so preview might not work with FLV anymore, please use the **HTML5** preview link instead:*

Multicast URL:Disable

SRT URL:srt://192.168.1.168:9000

SRT PUSH URL:Disable

Preview(HTML5) Preview(FLASH)

Note: meanwhile **we have erased the Flash-item from the web** and it is only based on HTML5 now:

Main stream

Encode Type:H.265

Encoding Type:1920x1080@25

Bitrate(kbit):6000

TS URL:http://192.168.1.168/0.ts   http://192.168.1.168:8086/0.ts

HLS URL:Disable

FLV URL:http://192.168.1.168/0.flv   http://192.168.1.168:8086/0.flv

RTSP URL:rtsp://192.168.1.168/0   rtsp://192.168.1.168:8554/0

RTMP URL: Disable

RTMP(S) PUSH URL: Disable

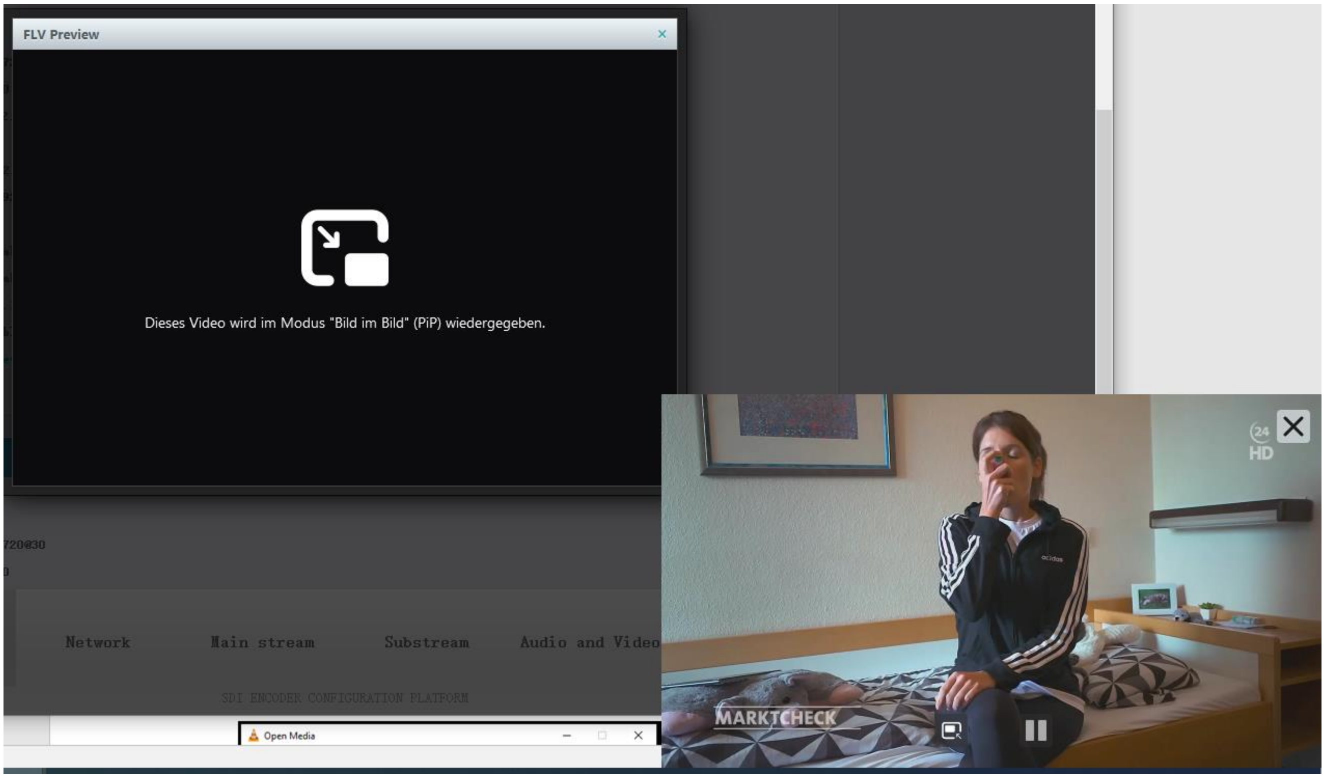
Multicast URL:Disable

SRT URL:Disable

SRT PUSH URL:Disable

Preview(HTML5)

-> HDMI-Input stream- Preview:



A PiP can be assigned as permanent preview in front to all other desktops.

Back to STATUS page:

Like the hint above, sometimes it's helpful to reload the Status page i.e. if you see @0:

to gather the actual values like Input HDMI values:

Running Time:0000-00-00 00:04:59

Device Time:2019-12-02 15:01:11(Sync Time To Device)

CPU Usage:11% (If CPU usage always more than 85%, please close some stream.)

Memory Usage:30.4M/485.6M

Input Size:1920x1080p@50

Collected Video Frames:14564

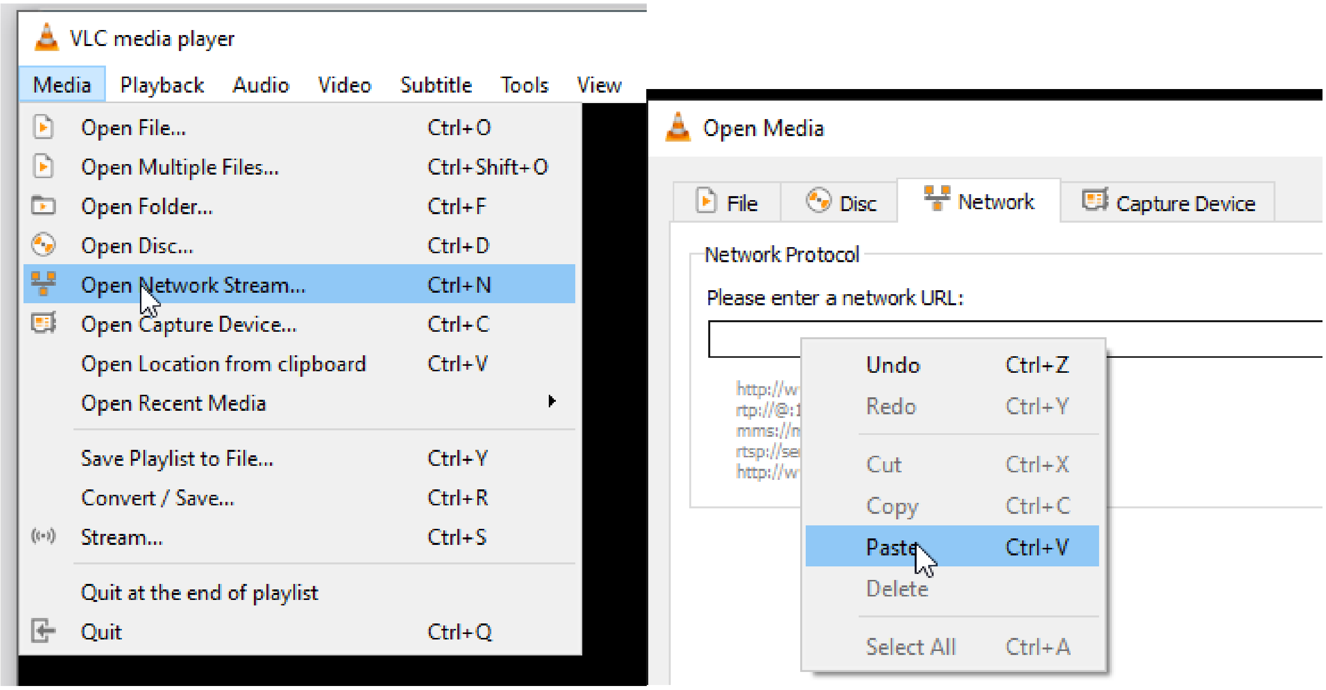
The device time can be adjusted by the Network-setup-part NTP-Server which you need to tell the NTP server URI and UTC-time difference. UK = '0', Germany normal is UTC+1...

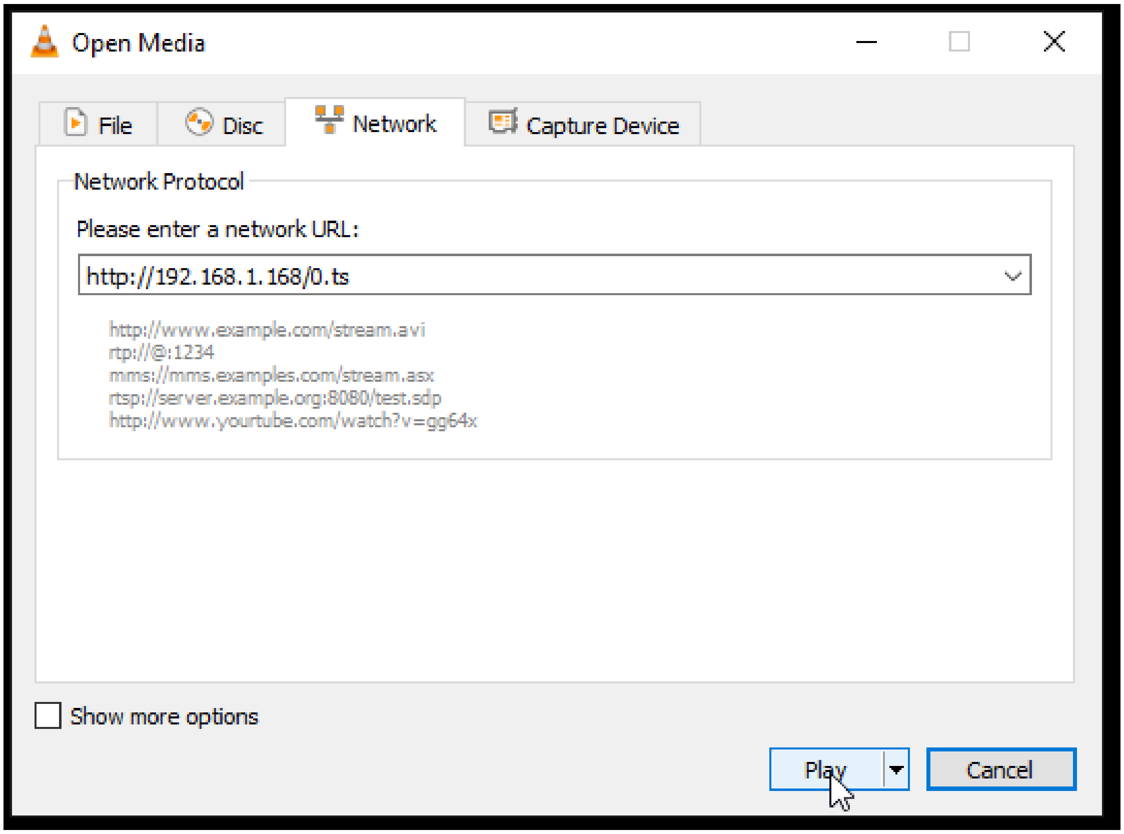
If you press (Sync Time to Device) it will be updated.

To also check your encoding streams, you can copy the URI from the STATUS page:

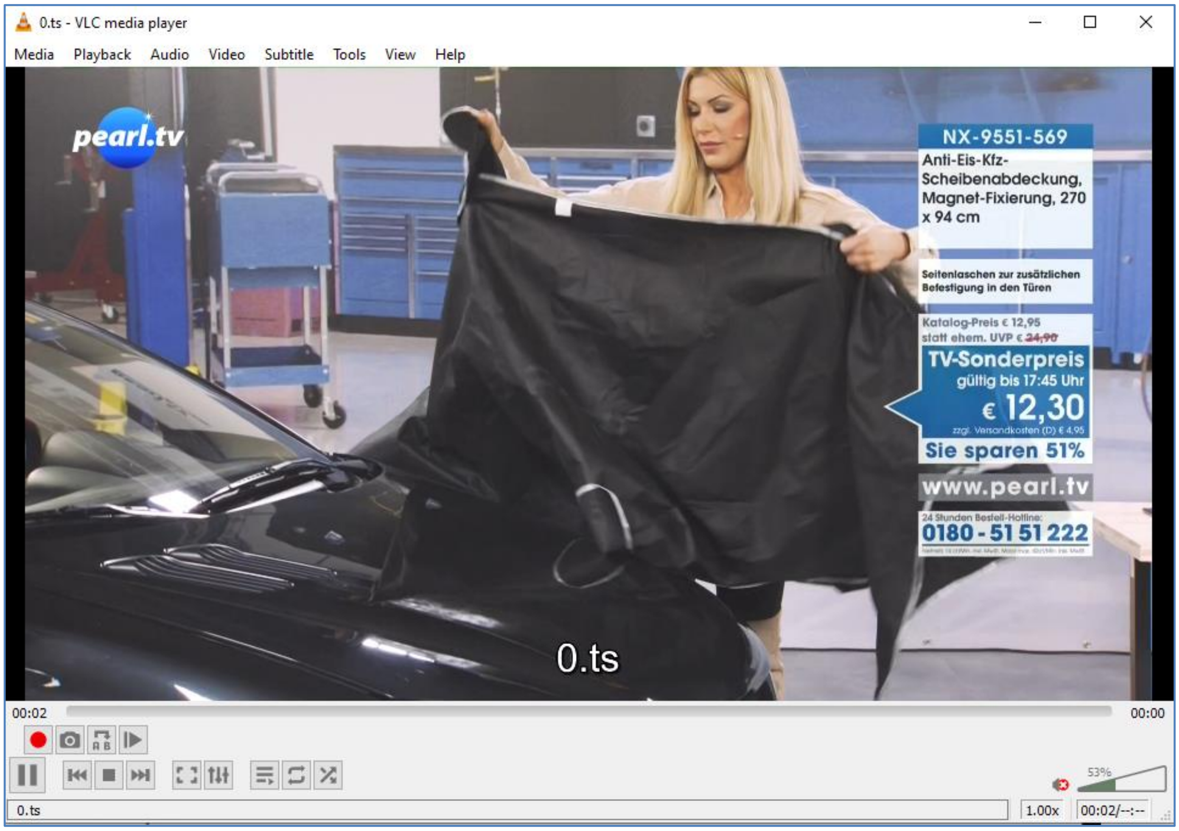
```
Encode Type:H.264

Encoding Size:1920x1080@25

Bitrate(kbit):2500

TS URL:http://192.168.1.168/0.ts    http://192.168.1.168:8080/0.ts

HLS URL:Disable

FLV URL:http://192.168.1.168/0.flv    http://192.168.1.168:8080/0.flv

RTSP URL:rtsp://192.168.1.168/0    rtsp://192.168.1.168:8554/0

RTMP PUBLISH URL:Disable

Multicast URL:Disable

Preview(Delay 1000ms)
```

Mark it by the mouse and COPY it - Than insert into VLC:

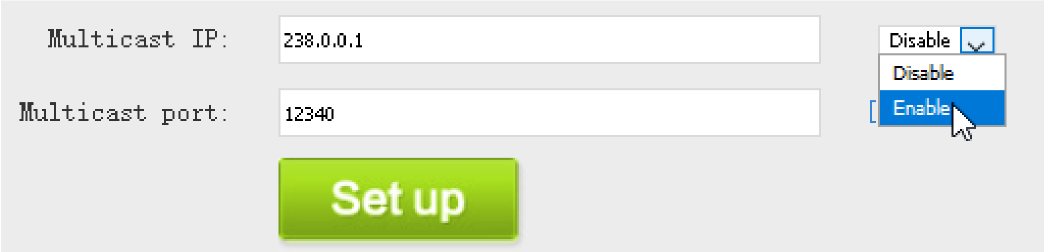


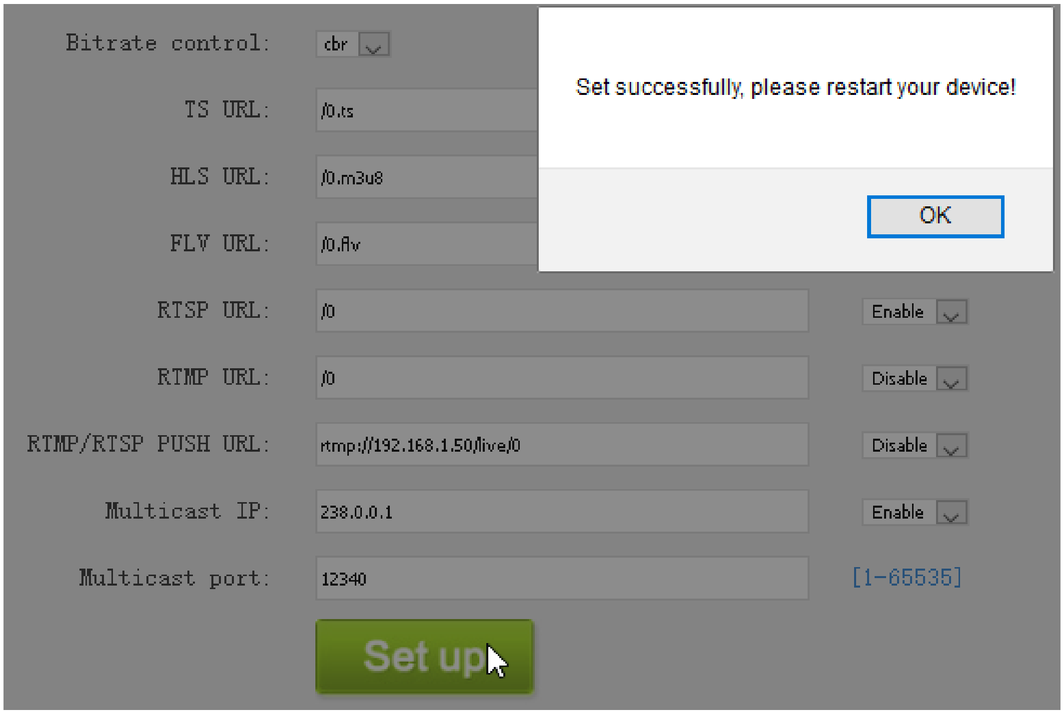


Note: If you more than one Network-Card in operation (like WIFI and GbE) in your receiving machine, VLC often does not recognize where to catch it from. Manually settings of METRIC Values for both can solve this issue.



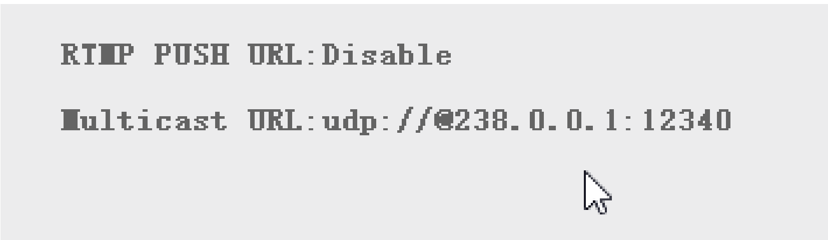
Note: UDP/RTP-Address will be taken by VLC with an @ and we have made it easy for you:

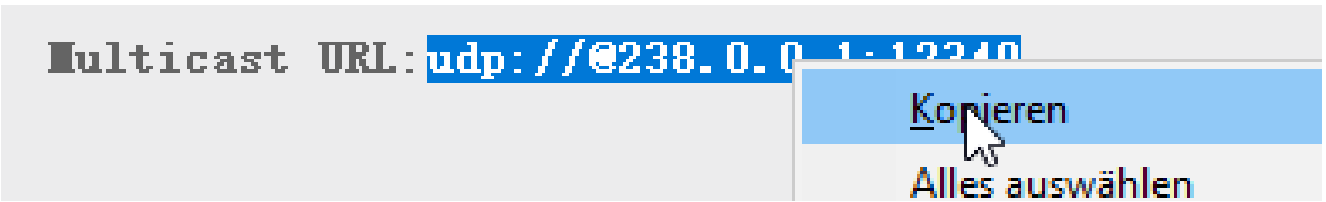


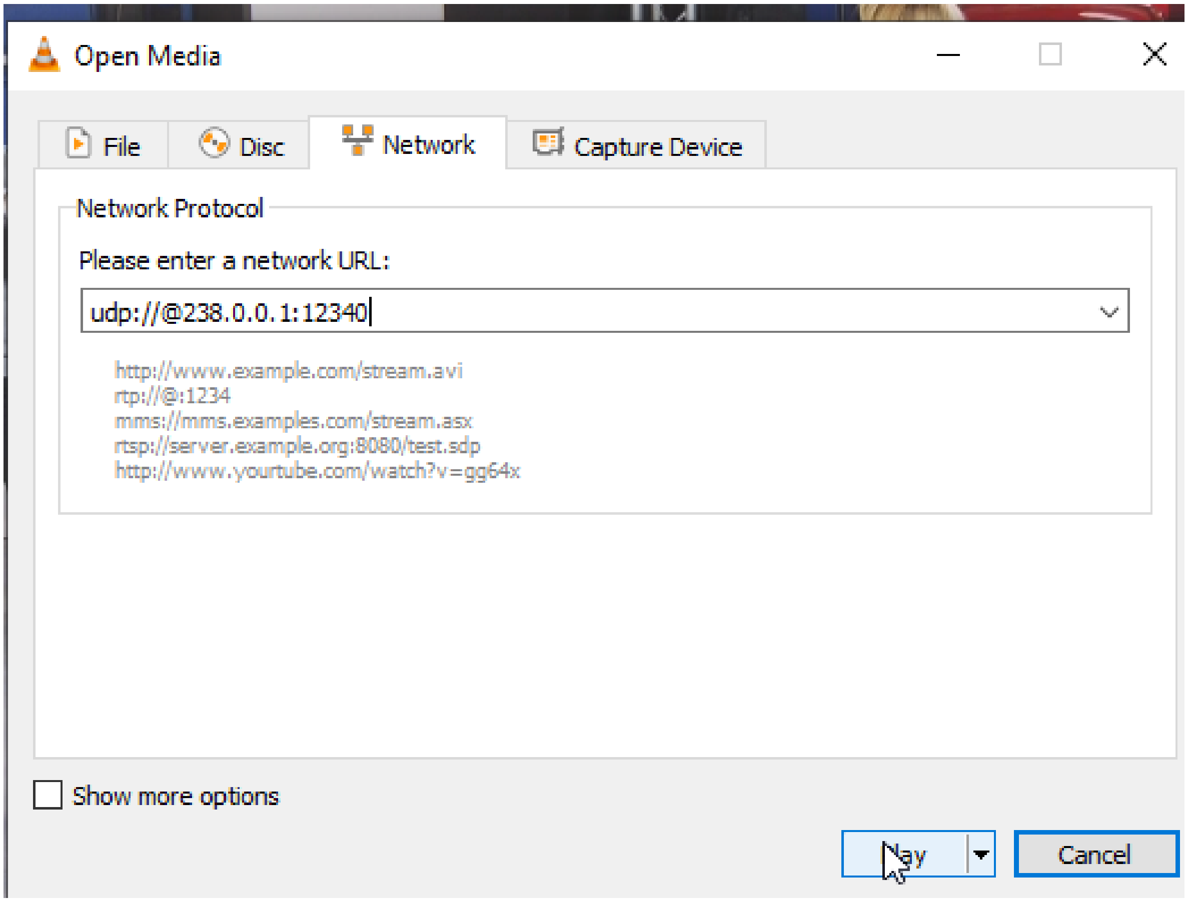


AGAIN: **You do not need to restart the encoder only the receivers you have in your network need to re-sync** to the changed values!!!

Multicasts:



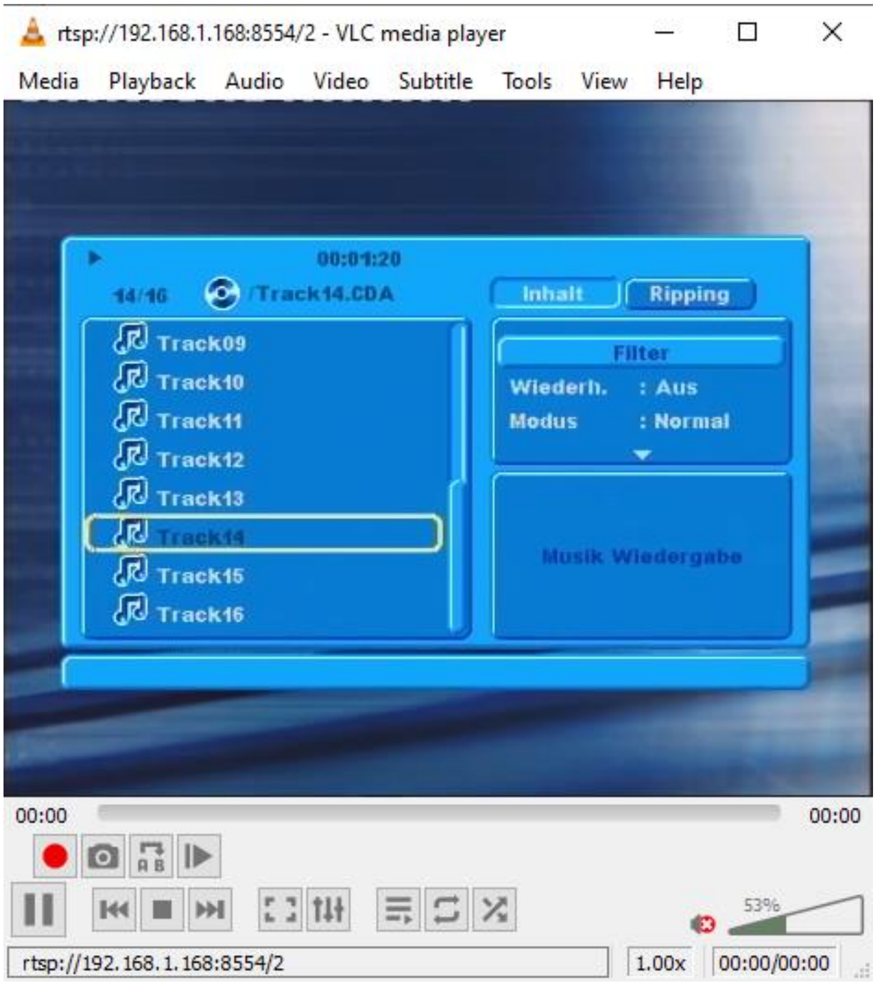



Open Media
File
Disc
Network
Capture Device
Network Protocol
Please enter a network URL:
udp://@238.0.0.1:12340
http://www.example.com/stream.avi
rtp://@:1234
mms://mms.examples.com/stream.asx
rtsp://server.example.org:8080/test.sdp
http://www.yourtube.com/watch?v=gg64x
Show more options
Cancel


udp://238.0.0.1:12340 - VLC media player
Media Playback Audio Video Subtitle Tools View Help
pearl.tv
NX-9397-569
Elektrische Kfz-Zusatzheizung, 12 Volt, Kugelkopf-Halterung, 130 Watt
Integrierte LED-Arbeitslampe
Katalog-Preis € 15,95
TV-Sonderpreis
gültig bis 17:51 Uhr
€ 15,15
Sie sparen 44%
www.pearl.tv
0180 - 51 51 222
udp://238.0.0.1:12340
00:00
00:00
53%
udp://238.0.0.1:12340
1.00x
00:00/00:00

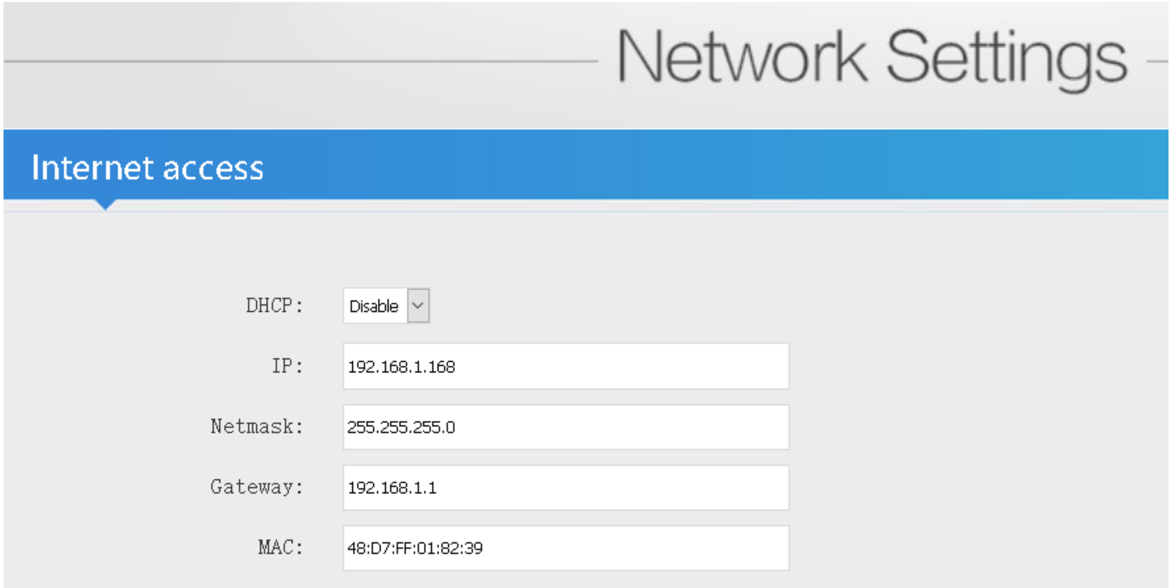
CVBS-Stream:

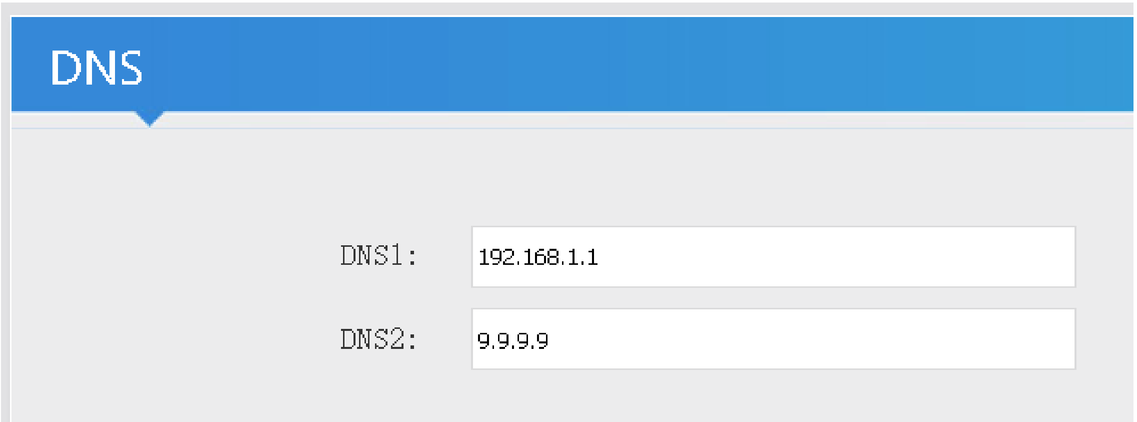
(Only ADE-Versions)



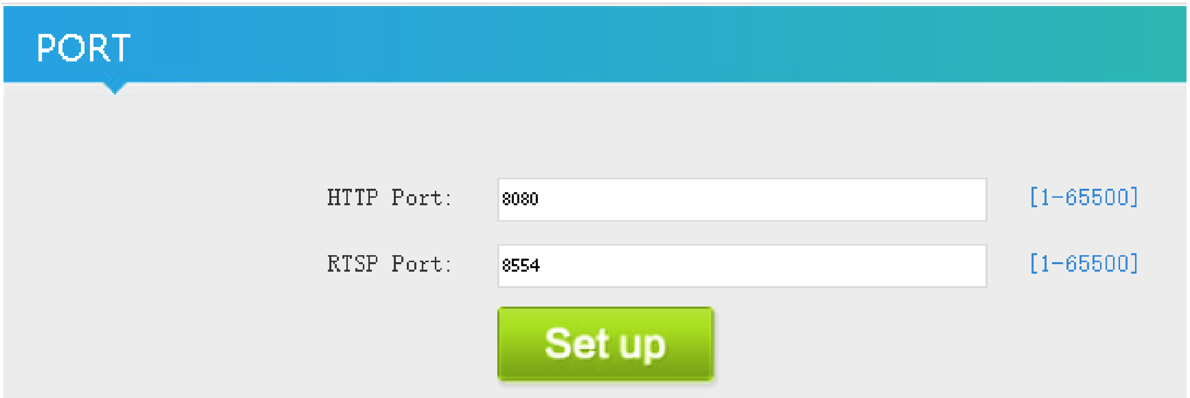
Network: **Here you can change the encoders IP-address and mode:**

If you change it to DHCP – after a reboot it will catch it from your router. Disadvantage: You need to check the the encoder given IP Address by your router in its own menu or use an IP-Scanner-tool.



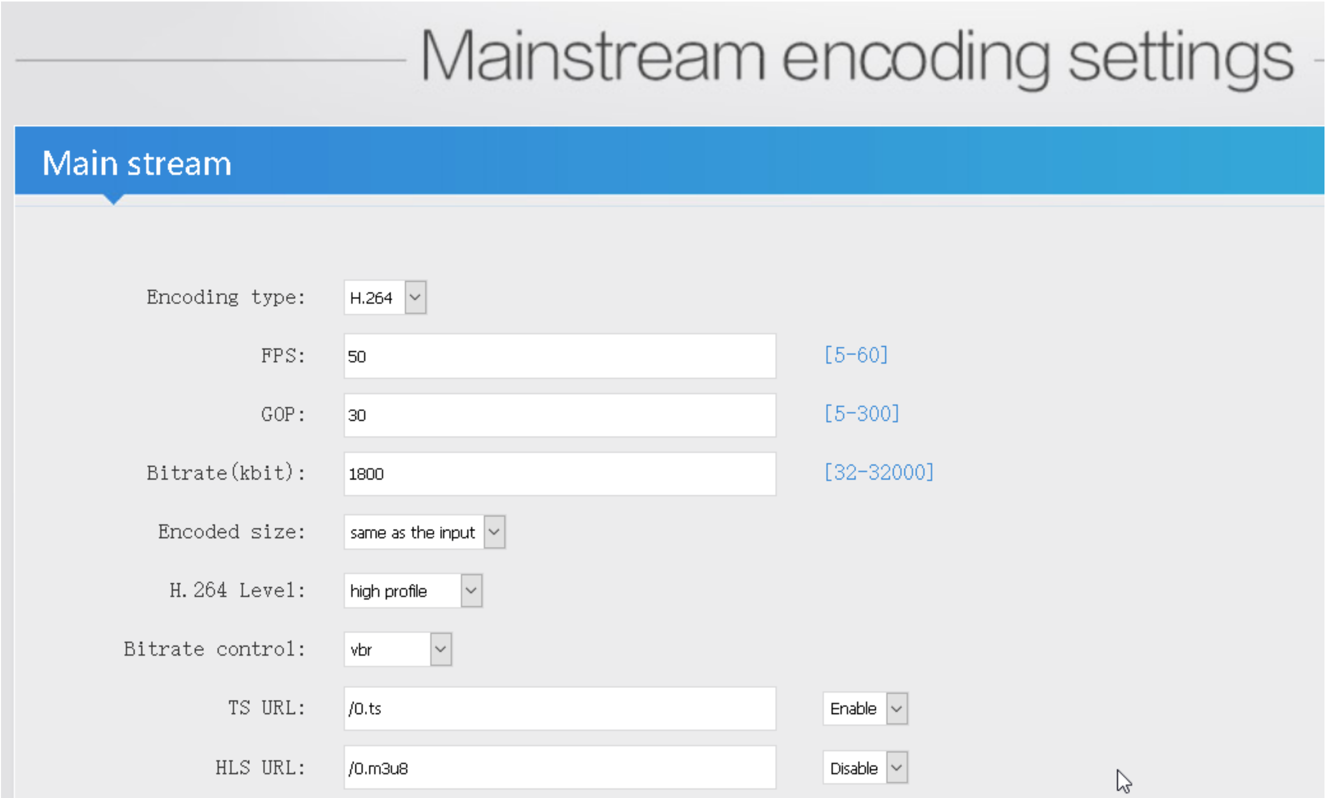


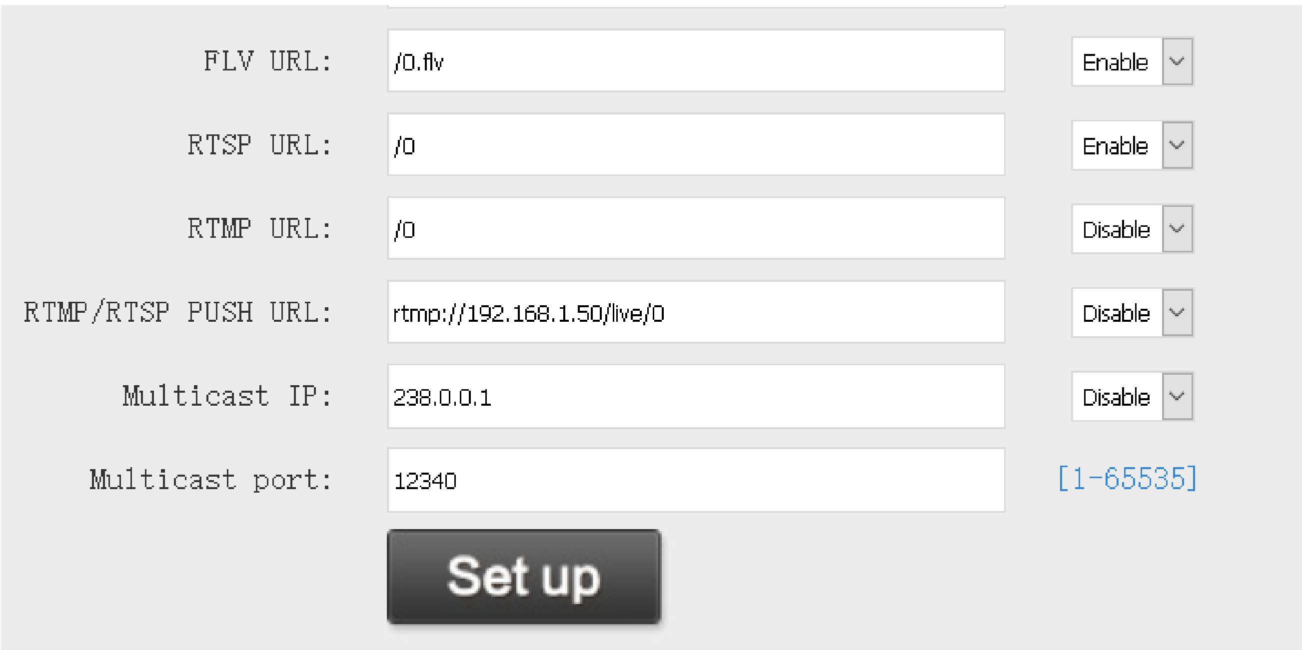
We assume that you are familiar with the basic settings of a network.



These are the basic ports for HTTP and RTSP-Streaming use. You can modify that but we recommend to keep them as they are because RTSP – receivers might are fixed to that port while HTTP is not. The bottom of the every of the menu-pages contain the 'Set up' buttons to take and enable your changes.

The **MAIN and SUB-Stream adjustments** are nearly all similar:

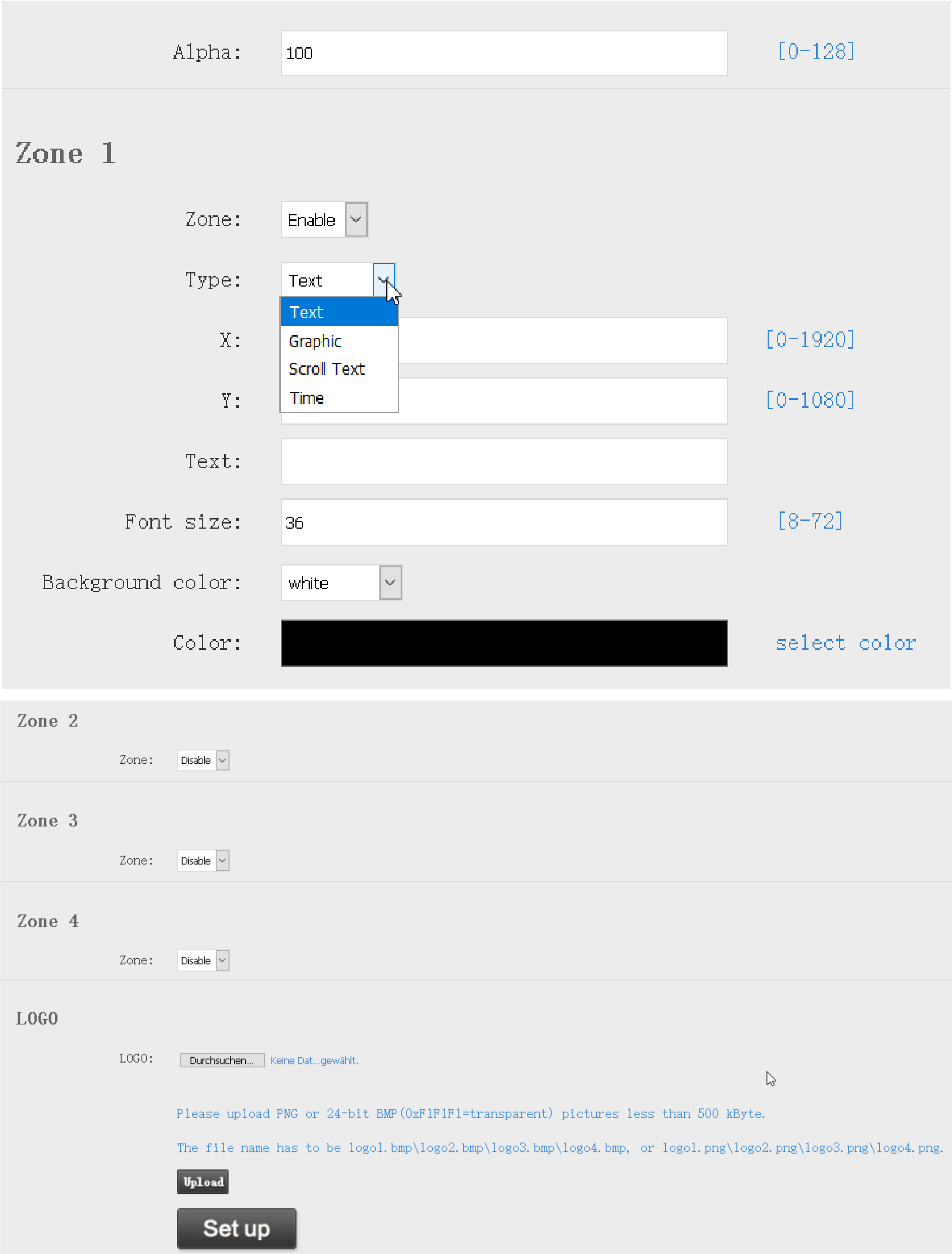




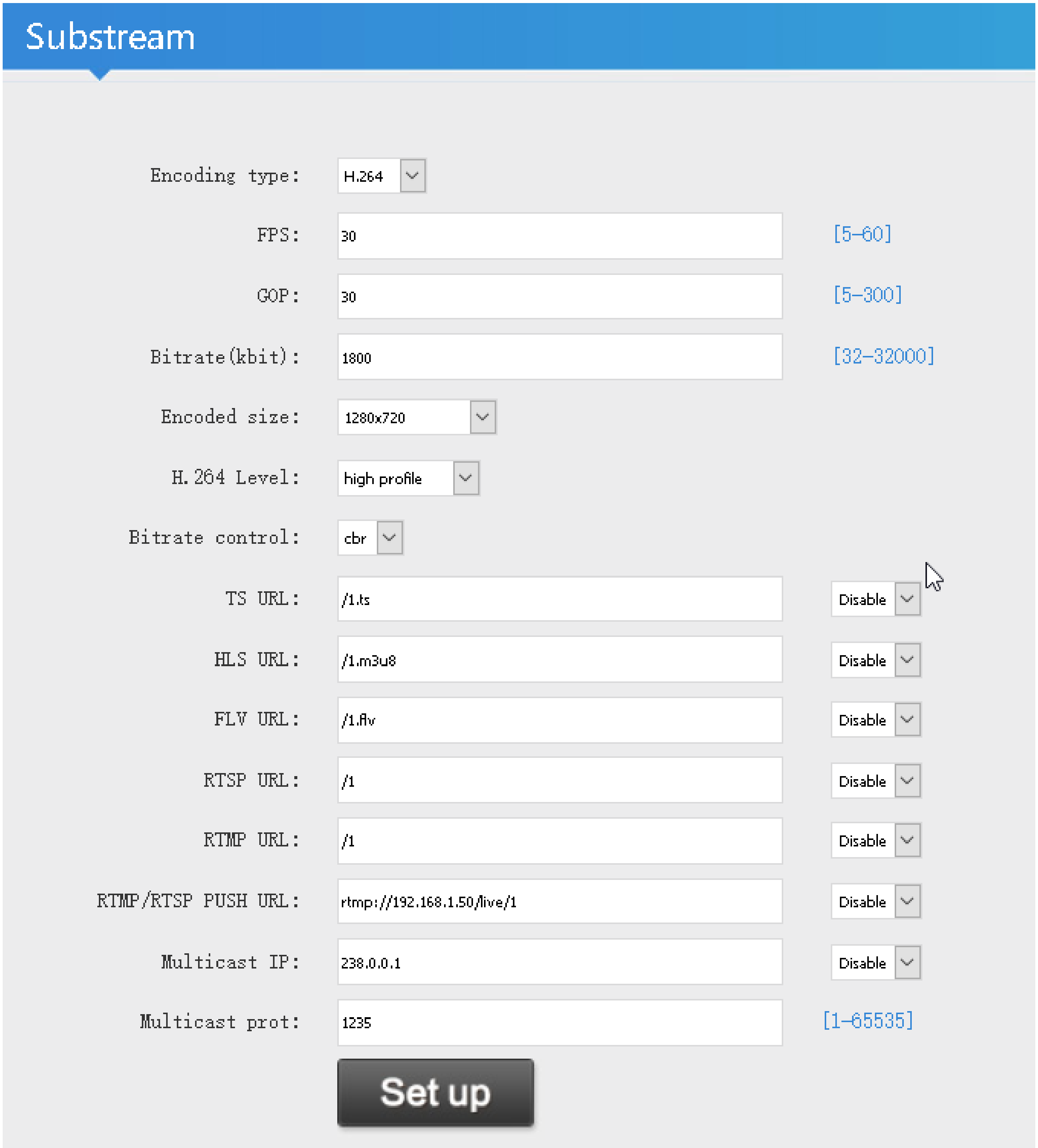
On Screen Display Menu: You can 'Overlay' a Text or Logo over the encoded Picture in 4 Zones:

For deeper detailed explanations about the OSD feature, refer to the full – Manual please.

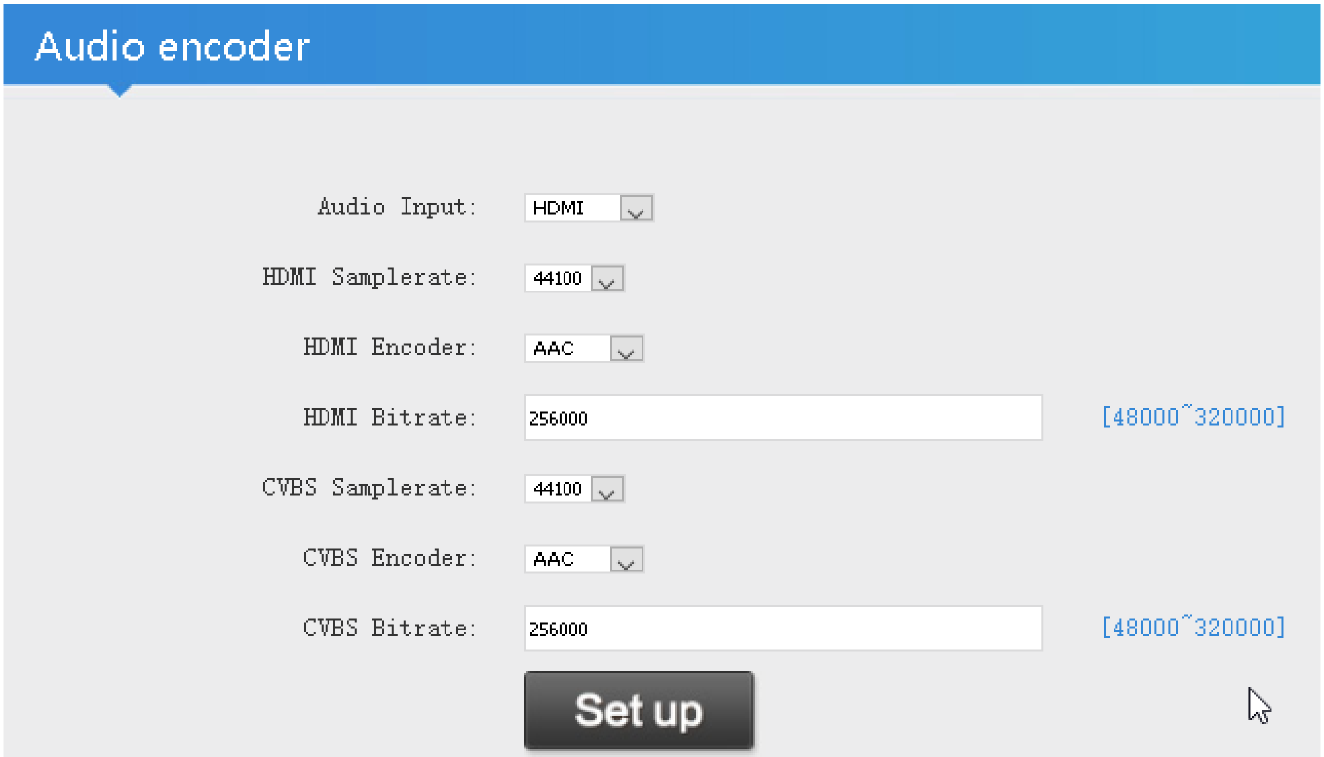
Also, for the ONVIF settings with RTSP.



It supports BMP with a special background colour if you like to be that transparent – or simply use already transparent PNG files. Names and limitations of size are shown in the web.

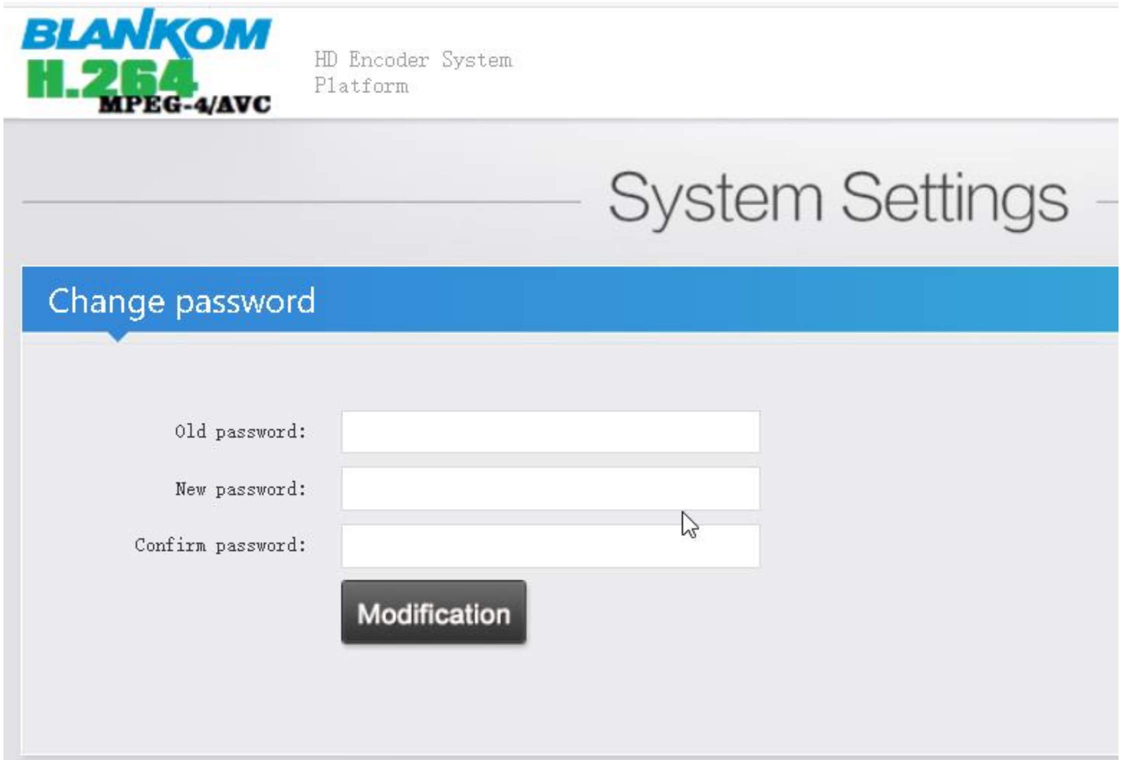

Substream
Encoding type:
H.264
FPS:
30
[5-60]
GOP:
30
[5-300]
Bitrate(kbit):
1800
[32-32000]
Encoded size:
1280x720
H.264 Level:
high profile
Bitrate control:
cbr
TS URL:
/1.ts
Disable
HLS URL:
/1.m3u8
Disable
FLV URL:
/1.flv
Disable
RTSP URL:
/1
Disable
RTMP URL:
/1
Disable
RTMP/RTSP PUSH URL:
rtmp://192.168.1.50/live/1
Disable
Multicast IP:
238.0.0.1
Disable
Multicast prot:
1235
[1-65535]
Set up

Audio settings are common for both stream encoder parts:

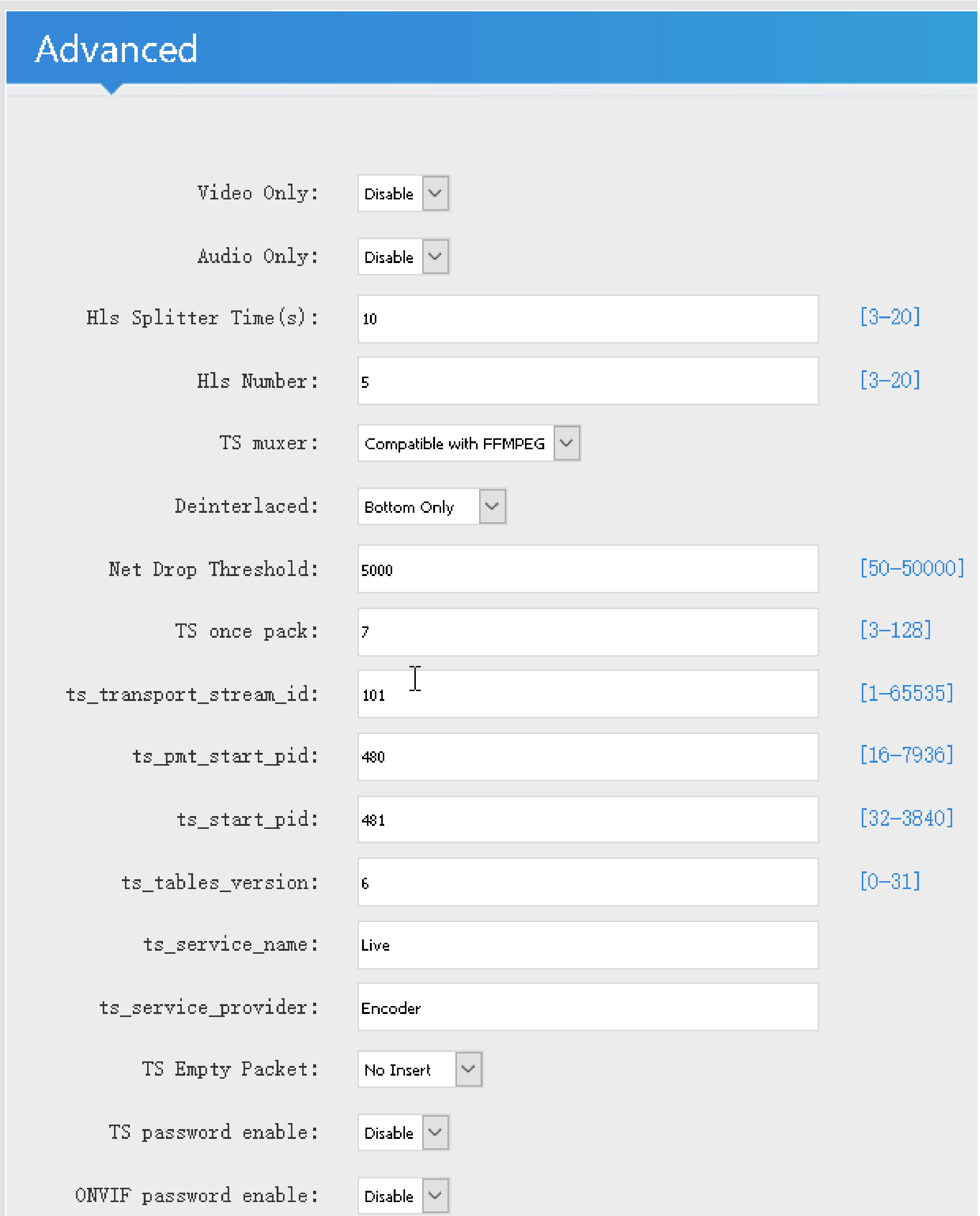


Add-on: Some models have OPUS support….

Self-explaining:



The default settings are usually Ok for most use-cases:



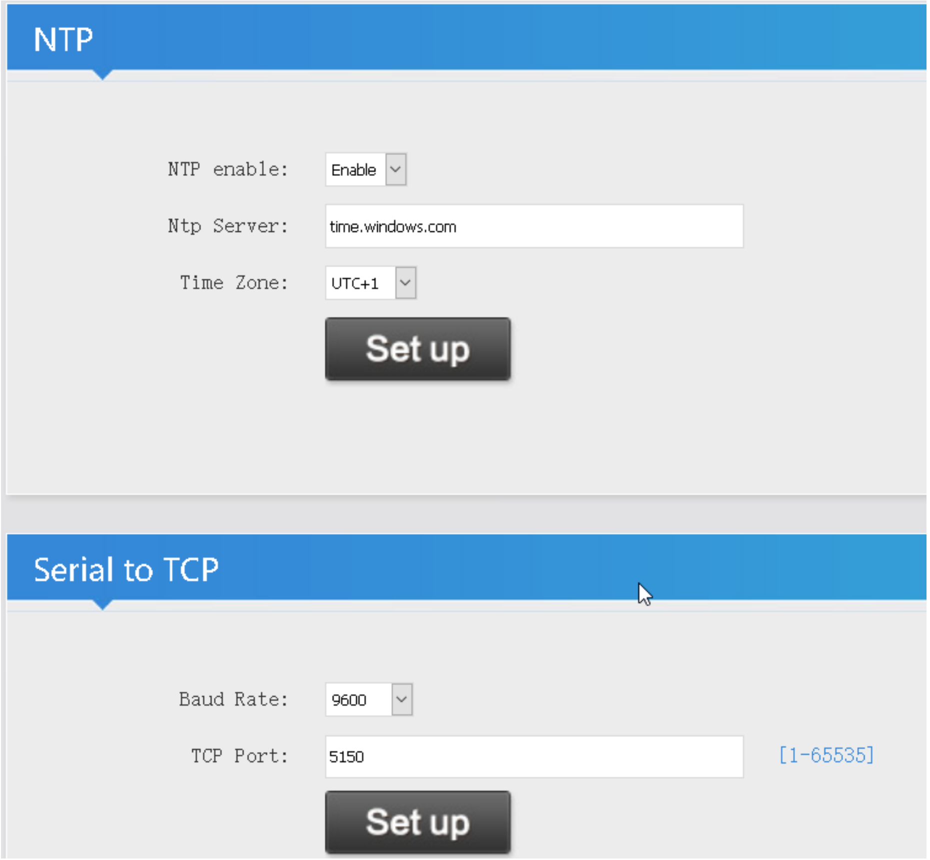
Playing with 'DE-interlaced settings' helps sometimes fixing moving picture artefacts.

BOTTOM only can solve right-left-camera moving sticking problems.

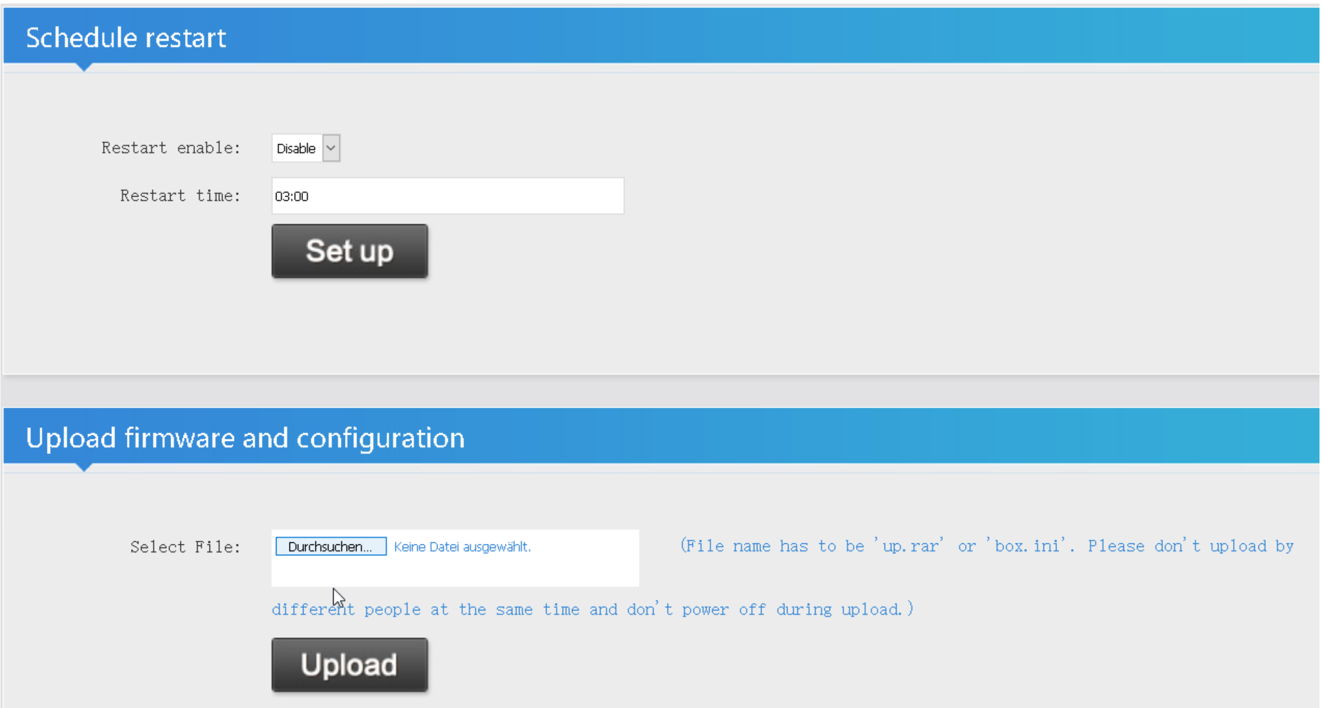
Vmix Compatible:	Disable	
TS OVER RTSP:	ES	
Multicast type:	UDP	
UDP TTL:	64	[1-254]
UDP SOCKET_BUF_SIZE:	20971520	(0-20971520]
Slice split enable:	Disable	
Slice size:	1024	[128-65535]
MIN_QP:	5	[1-35]
MAX_QP:	42	(MIN_QP-50]
	Set up	

For more info... contact us... www.blankom.de

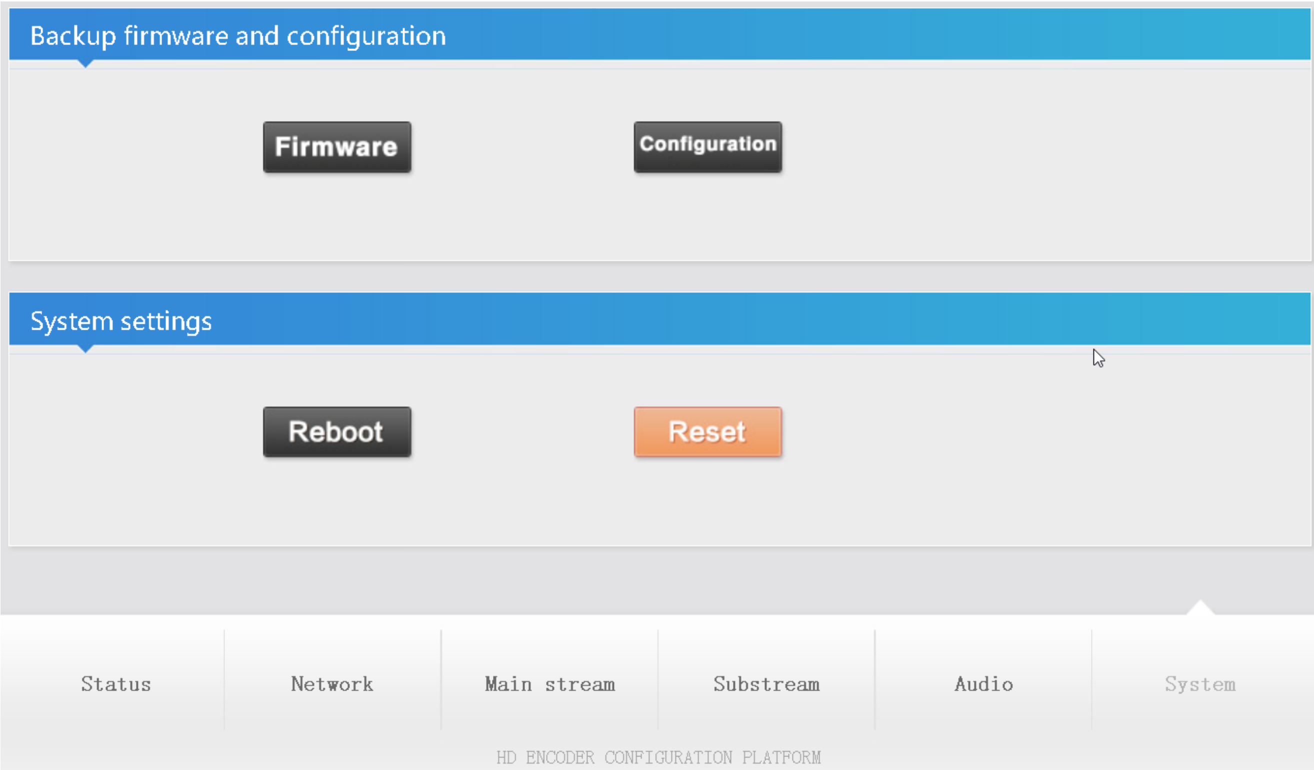
A schedule 'restart' can be programmed (NTP-Time = ON recommended):



Supporting **Rserial** function if needed (Linux like)

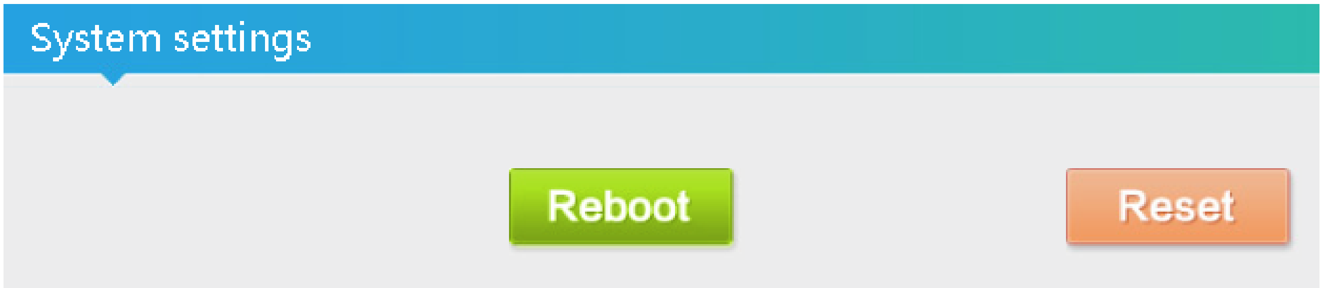


The settings as well as the Firmware can be back-upped and re-uploaded.



The config-settings file is a Linux based text file named box.ini. Do not modify / store upload that by a windows editor except you will use notepad++ (freeware – please google...)

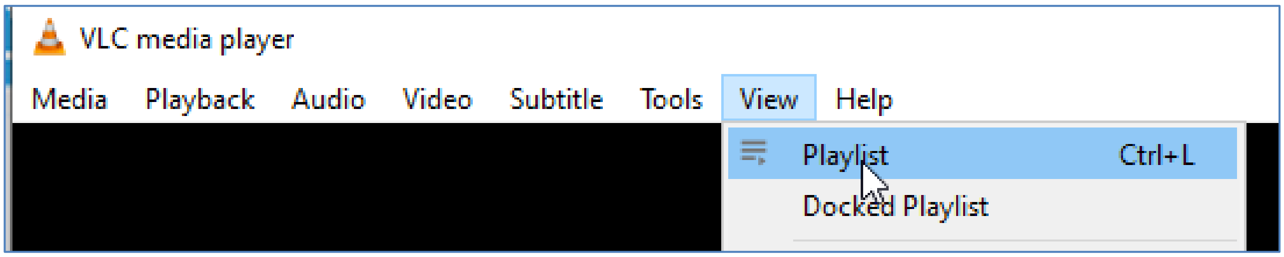
Finally, i.e. after firmware update has been uploaded, the unit can be remotely reset to factory defaults or rebooted:

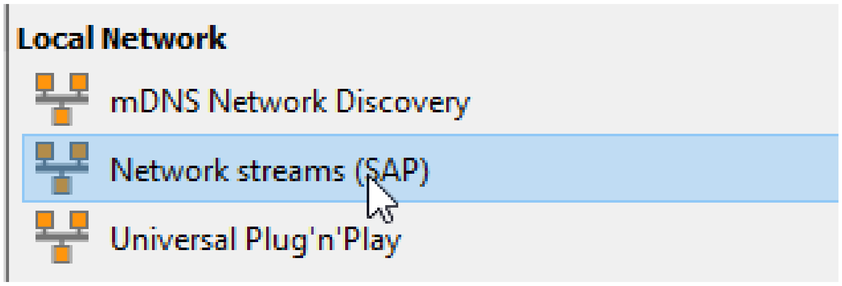


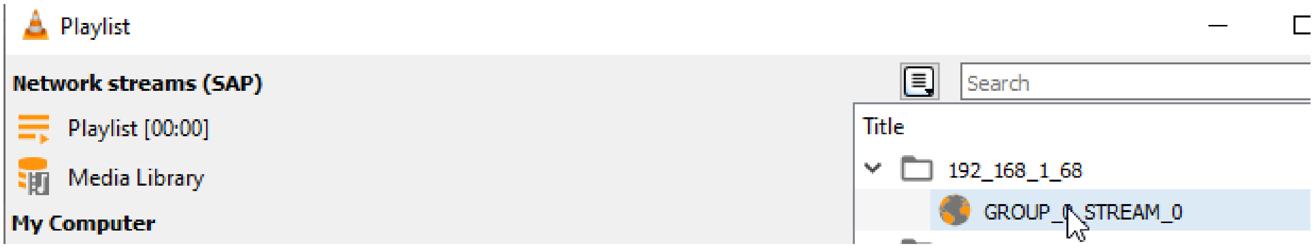
We recommend to make yourself familiar with 'What is Multicast and Unicast' and the corresponding IP-Ranges.

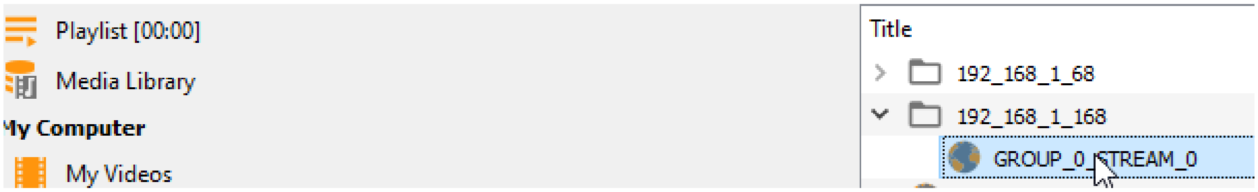
A last hint:

Using VLC SAP-Gathering will show a simple click'n start entry:





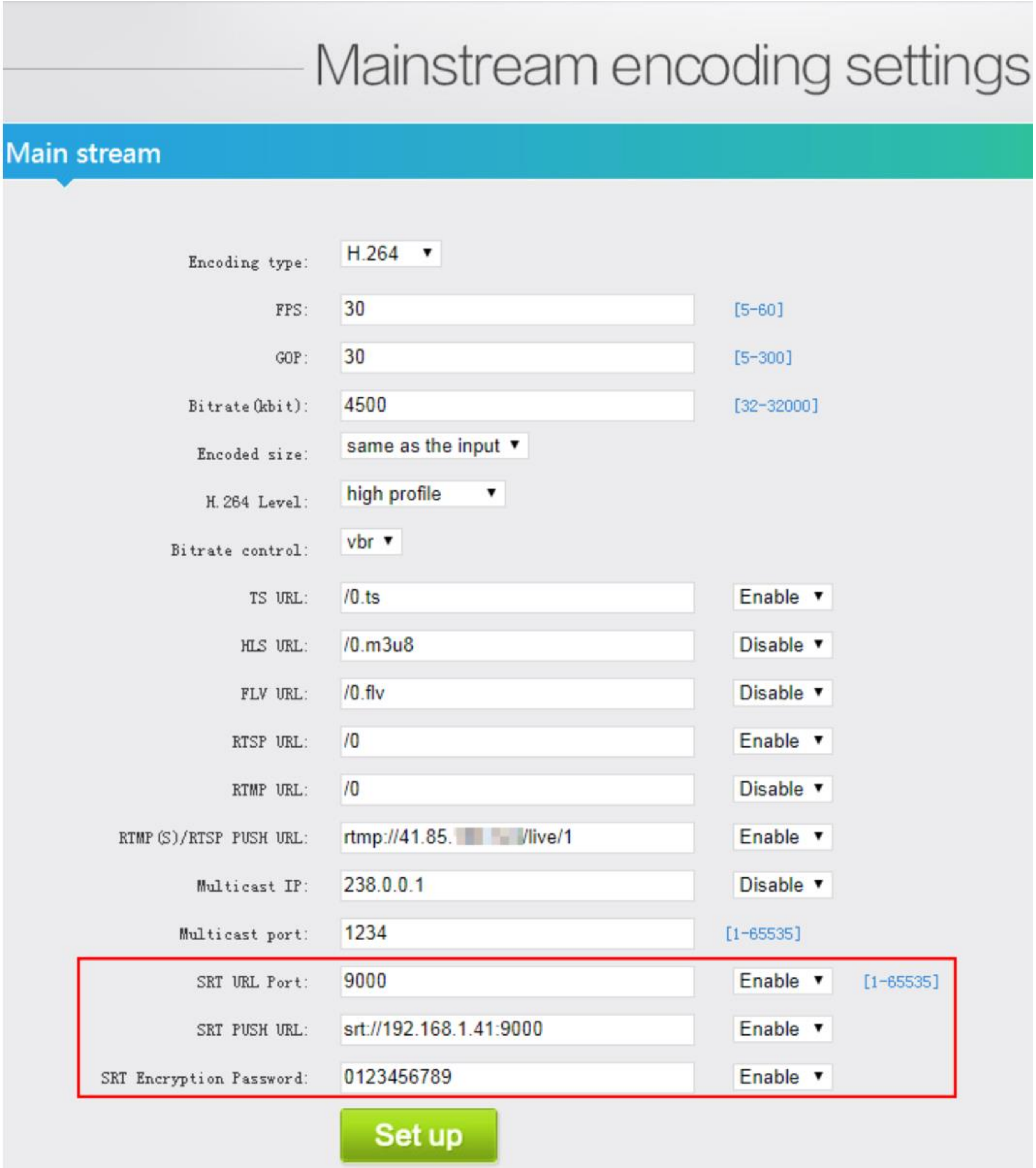




-> Will receive the stream. This works only with Multicast UDP / RTP !

Addon from FW 6.39 (03-2020-old hardware model) on:

SRT support: *(Only supported by our encoders with h.265 compatibility because of processing power)*



More details:

https://www.srtalliance.org

What is an SRT?

Secure Reliable Transport (SRT) is an Open-source software protocol and technology stack designed for live video streaming over the public internet.

SRT provides connection and control, reliable transmission similar to TCP, however, it does so at the application layer, using UDP protocol as an underlying transport layer. It supports packet recovery while maintaining low latency (default: 120 ms). SRT also supports encryption using AES.

Source: https://en.wikipedia.org/wiki/Secure_Reliable_Transport

Note: SRT works only in pairs: The stream receiver must support SRT reception.

Video Encoders are widely used in video transmission field, and SRT supported by our video encoder & decoder. Our Encoder & Decoder work perfectly for Haivision Play, Larix Broadcaster, etc.

srt-live-server(SLS)-for our Video Encoder

Our Video Encoder supports SLS for SRT.

Introduction

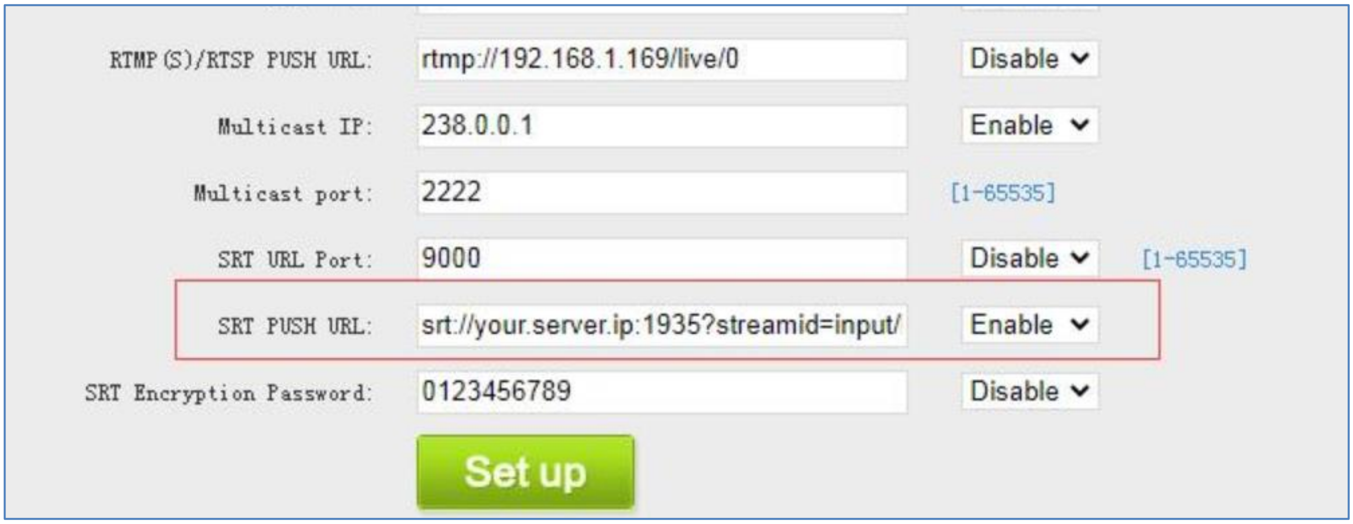
srt-live-server (SLS) is an open source live streaming server for low latency based on Secure Reliable Transport (SRT). Normally, the latency of transport by SLS is less than 1 second via the internet.

Requirements

Please install the SRT first, refer to SRT (https://github.com/Haivision/srt) for system environment basics. SLS can only run on OS based on linux, such as mac, centos or ubuntu etc.

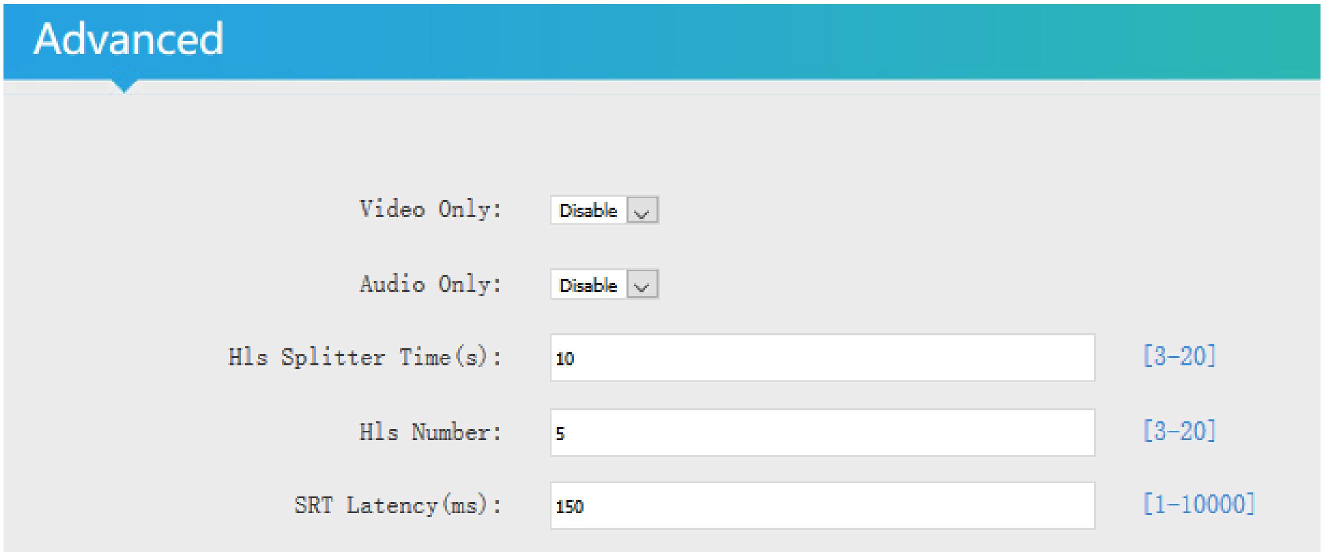
Source: https://github.com/Edward-Wu/srt-live-server

Put the following url to send to your docker container: srt://your.server.ip:1935?streamid=input/live/yourstreamname



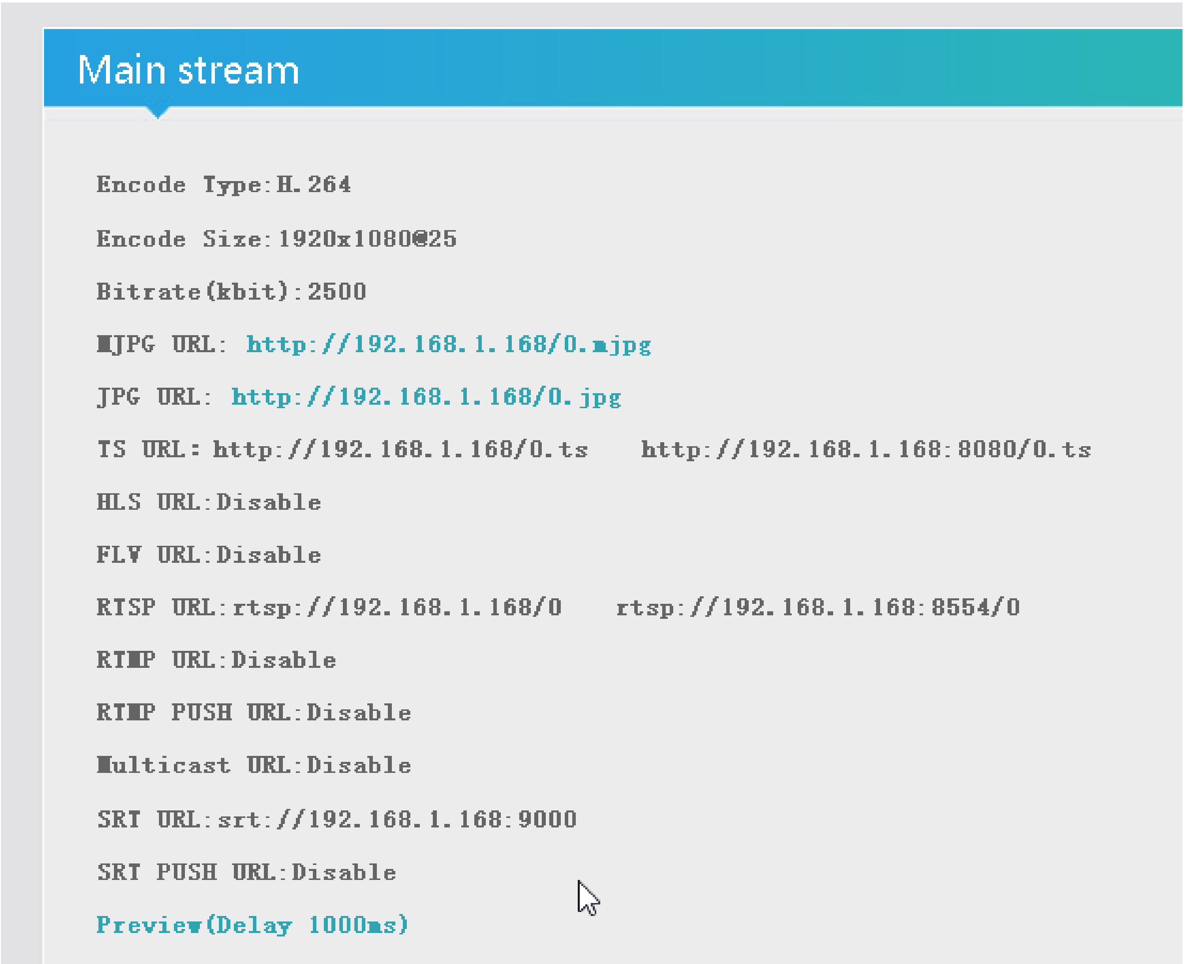
For P2P, select SRT PUSH and enter the destination IP Address and Port.

SRT network-Latency can be adjusted in SYSTEM *Firmware Version depending…* **:**



It is a faster transport protocol for lower latency over public networks…

Check the Status page:



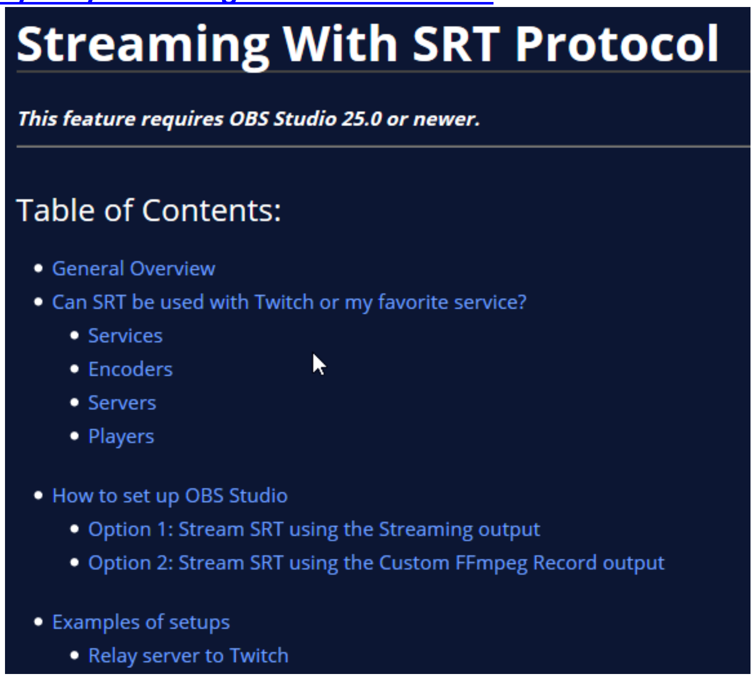
You can check the receiving it by a PC and VLC: *(please note, **the @** in the URI is not necessary like in udp/rtp)*

Some more useful links regarding SRT:

A Media server to handle SRT and more: The Open Broadcaster Software

https://obsproject.com/

https://obsproject.com/wiki/Streaming-With-SRT-Protocol:



Streaming With SRT Protocol

This feature requires OBS Studio 25.0 or newer.

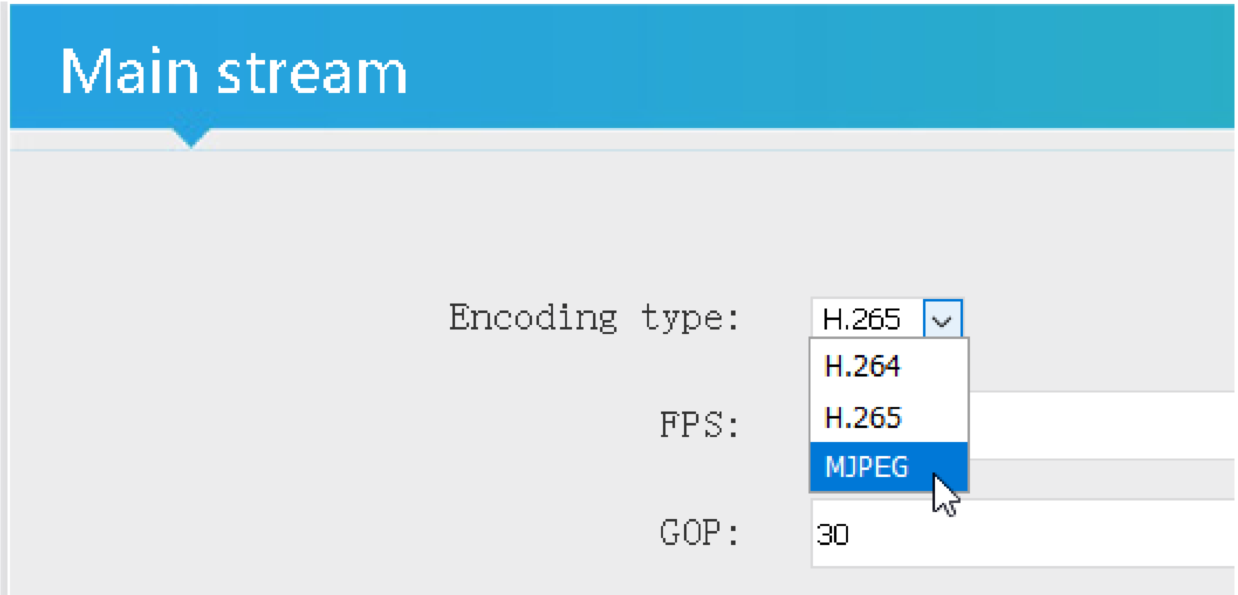
Table of Contents:

- General Overview
- Can SRT be used with Twitch or my favorite service?
  - Services
  - Encoders
  - Servers
  - Players
- How to set up OBS Studio
  - Option 1: Stream SRT using the Streaming output
  - Option 2: Stream SRT using the Custom FFmpeg Record output
- Examples of setups
  - Relay server to Twitch

https://github.com/obsproject/obs-studio
https://github.com/haivision/srt

MJPG Support:

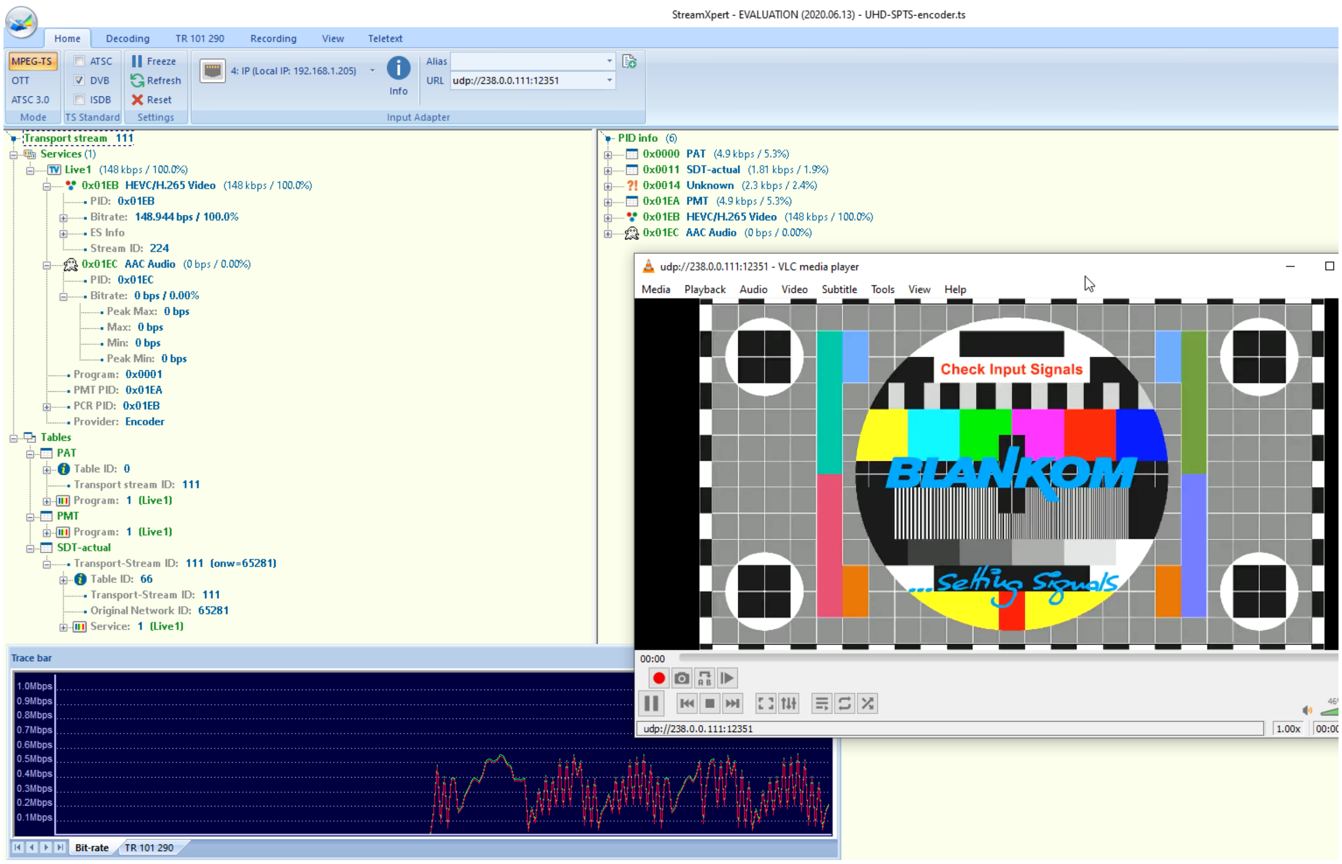
Setting the encoder main or secondary processor to



Enables at the Status-Page the direct Links for Motion JPEG transmission direct into your browser (if that supports it):

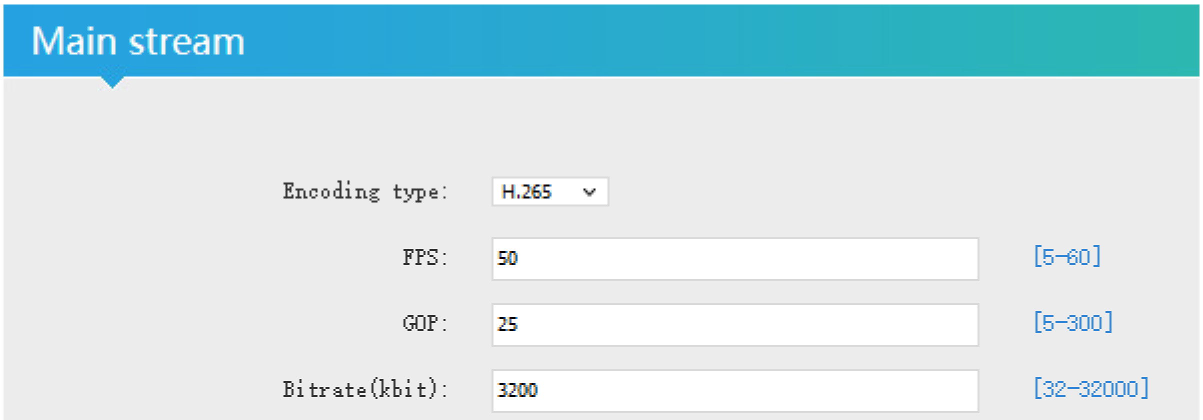
Just click: (here w/o input signal, so you get our test-picture ;-):

The /0.jpg (or in secondary stream the /1.jpg
Will do a screen-snapshot to your browser – so no motion – just like a screenshot.
BTW: If no signal has been detected at the Input connector, the Test-picture will appear and the Stream output may 'pump' because the encoder check the input signal periodically – and in this periods', the output stream might fluctuate like:



New feature added in Version 5.11:

- **HEVC h.265 Preview with inbuilt player (w/o pause/stop rew/fwd):**



Main stream

Encode Type:H.265

Encoding Type:1920x1080@50

Bitrate(kbit):3200

TS URL:http://192.168.1.168/0.ts   http://192.168.1.168:8086/0.ts

HLS URL:Disable

FLV URL:http://192.168.1.168/0.flv   http://192.168.1.168:8086/0.flv

RTSP URL:rtsp://192.168.1.168/0   rtsp://192.168.1.168:8554/0

RTMP URL: Disable

RTMP(S) PUSH URL: Disable

Multicast URL:Disable

SRT URL:Disable

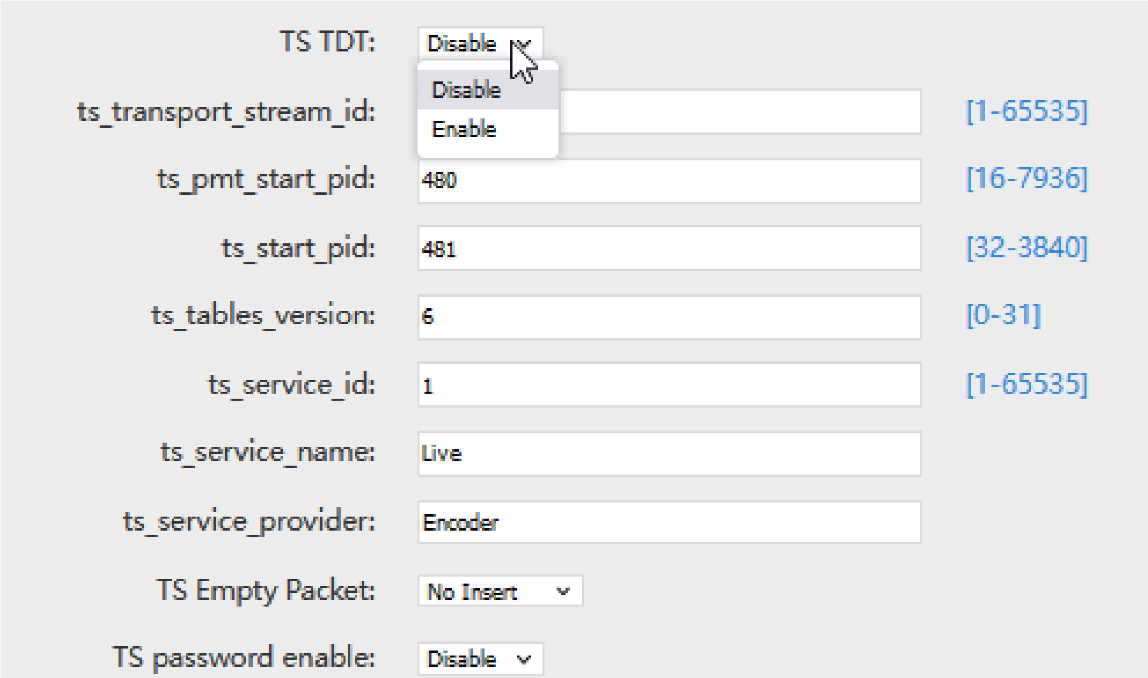
SRT PUSH URL:Disable

Preview(HTML5)

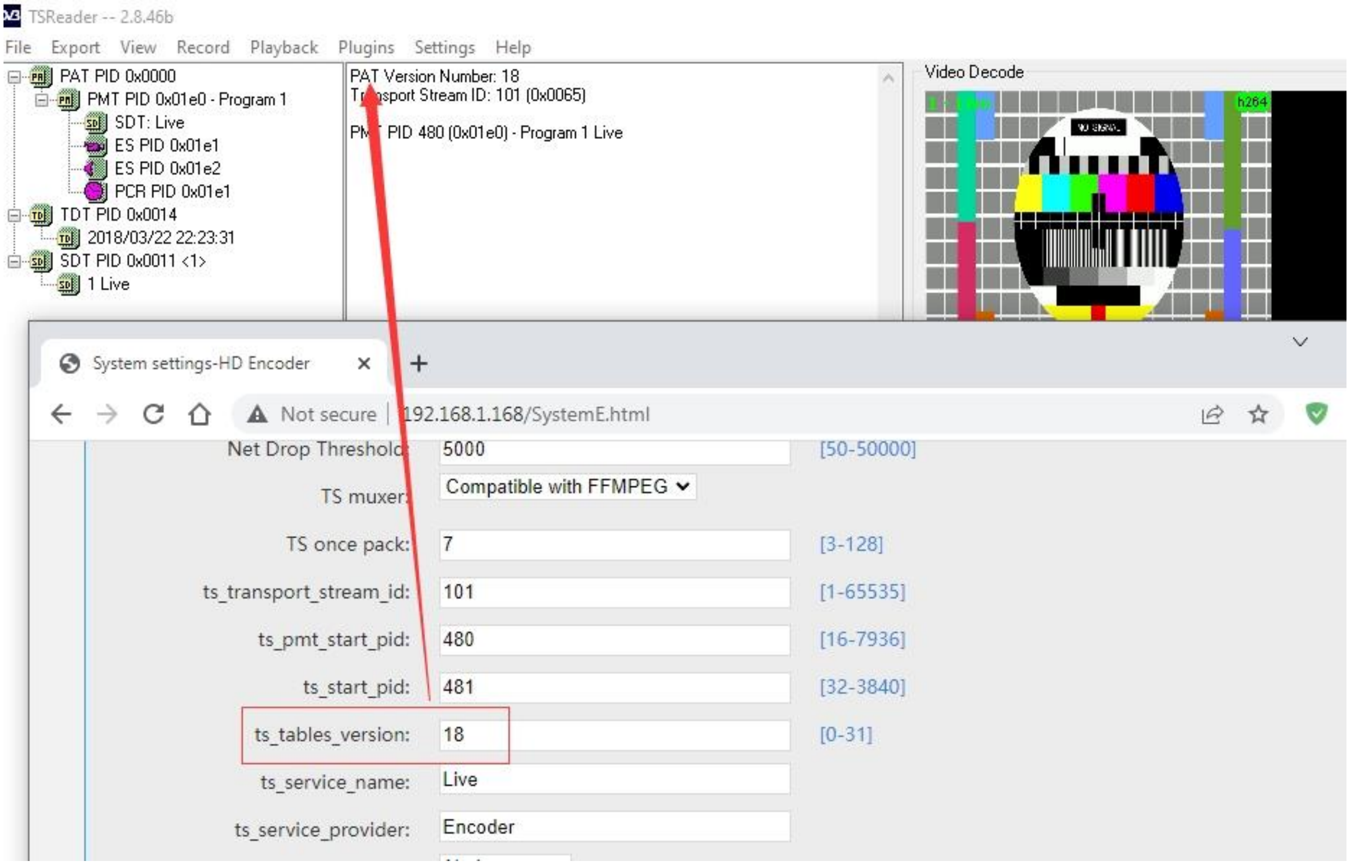
It can take several seconds until the preview starts but it highly depends on the receiving web-browser-PC hardware capabilities to decode that HEVC-PiP. So please be patient and wait a little…

Changing possibility of the Transport Stream-PID-ID-values to distinguish several encoders in a common network to finally use a multiplexer w/o PID-Remapping:

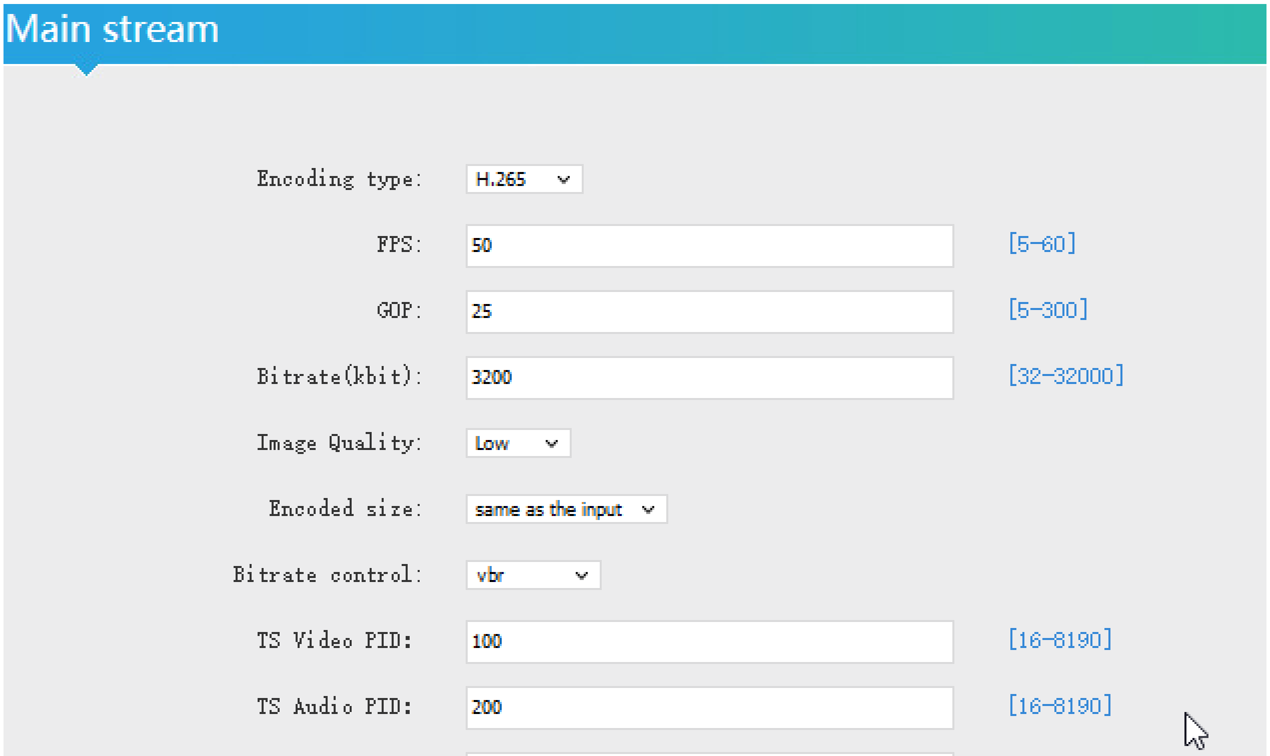
This in System-settings



TS Tables Version is related to the PAT (See MPEG-DVB):



In combination with:

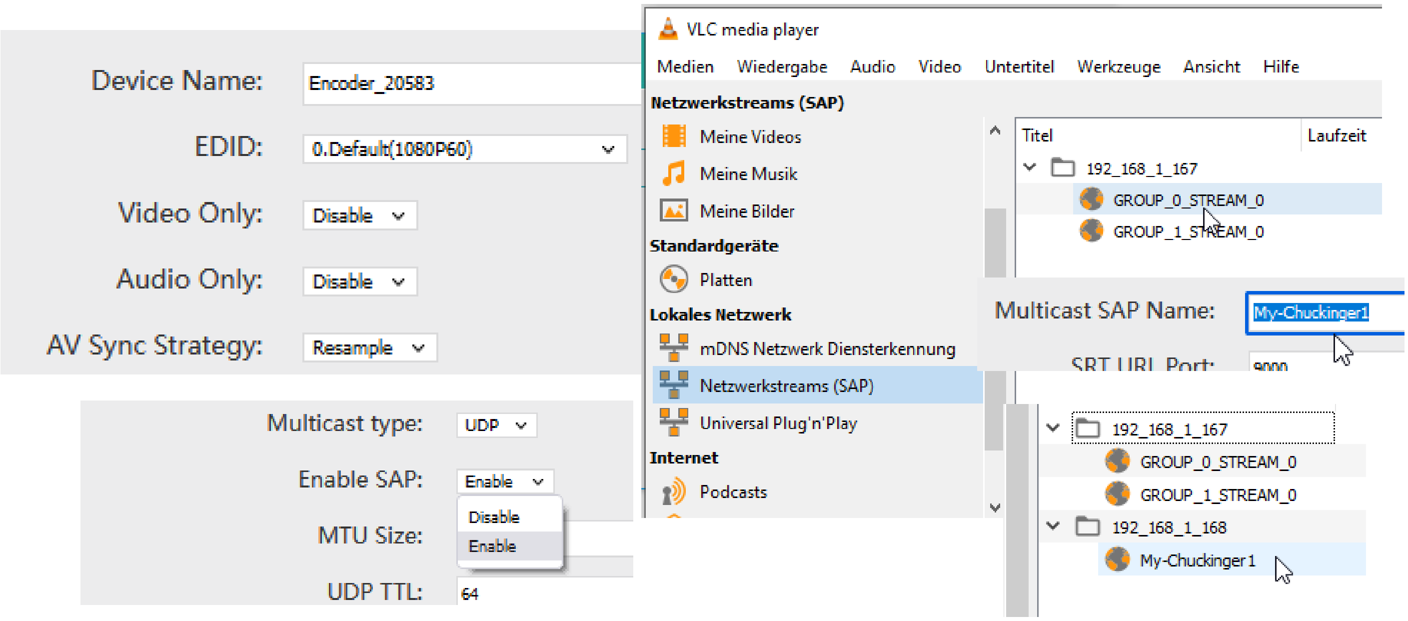


Please do not use PID's (here in Decimal instead of HEXadecimal in use) which are reserved in DVB, 0-18 are for special tables like PID 18= EIT. 8191dec is for Zero-fillings to a CBR TS.

Please check DVB-Norms if you are unsure.

SDE-265 and HDE-265L New Version 5.15…20 ADD ONs:

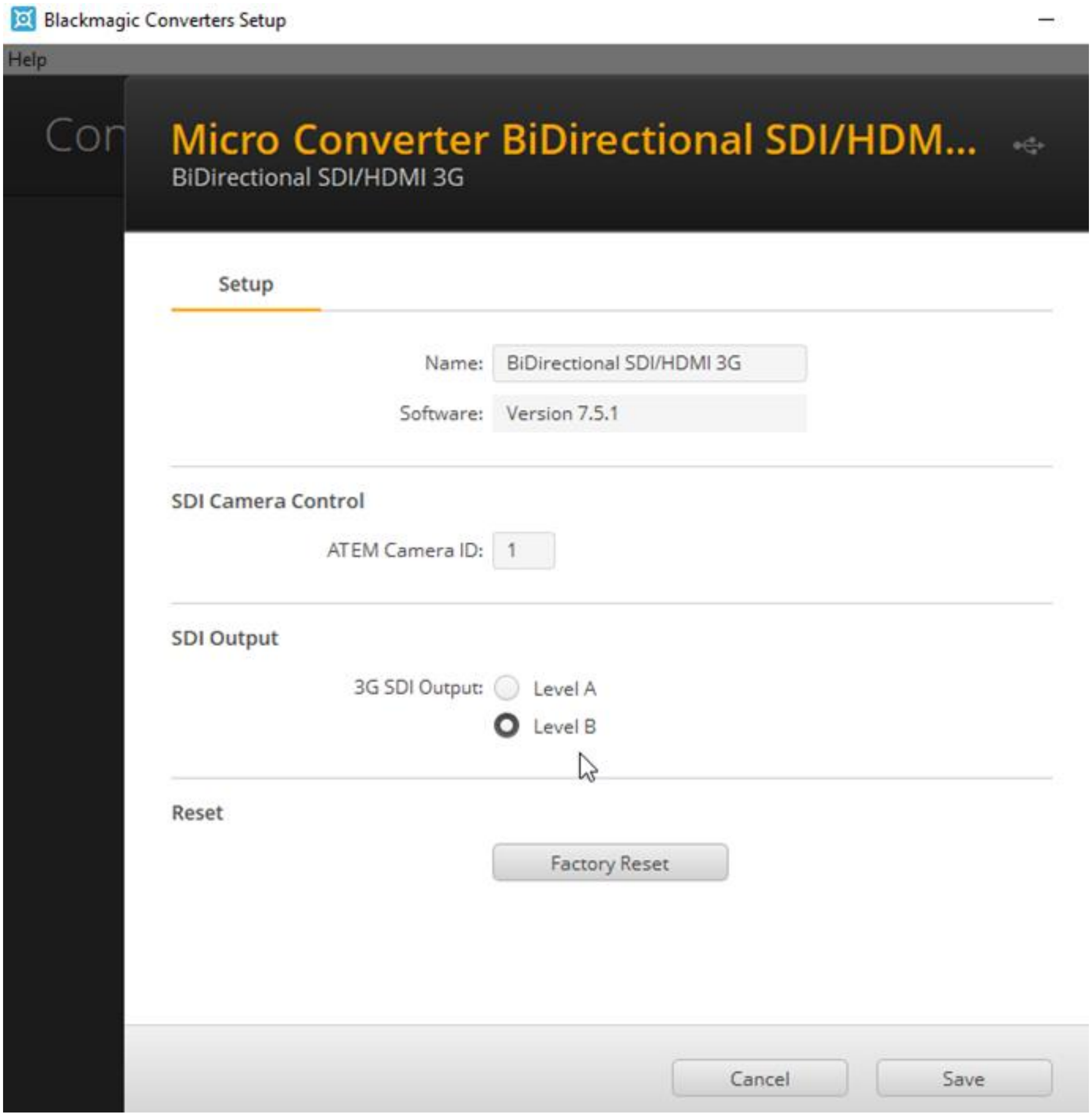
- New User Interface lookalike
- Inventing a Windows tool to search for your en- decoder if you lost IP Address: Find Your Encoder_Decoder.exe -> If you need that- ask us at info@blankom.de
- Changing possibility of TS Video & Audio PID and TS_service_id
- added RTSP- multicast support
- Changing of multicast stream SAP name option -> See below
- Added a checkbox/switch for TS TDT - System -> Advanced settings



Version 5.17 (April 2022, June 2022)

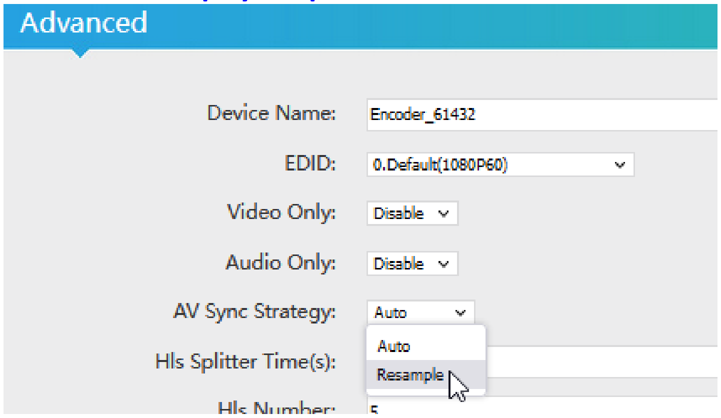
Step1: fixing Preview Window when HDMI contained no Audio signal

Step2: adds the SDI-Input detection of Level A and B automatically.
Because of some SDI-devices are using Level A and the SDE-265 needed Level B. Example:

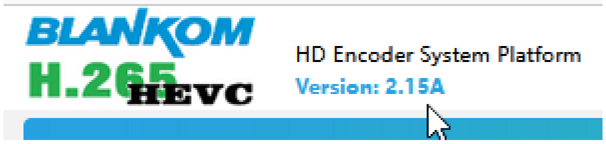


We added the automatic Level detection A and B for the SDI-Input.
(Not available in the old Hardware with firmware versions 6.xx)

Version 5.20: **lip sync option added:**

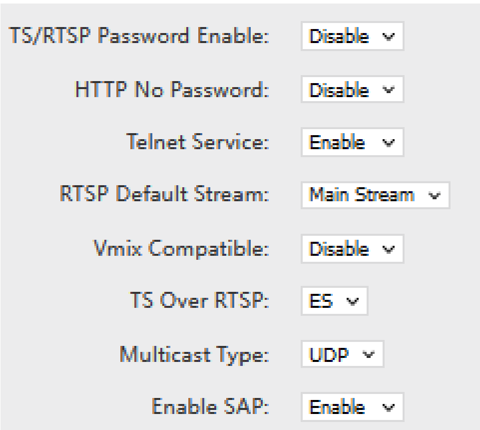


Intermediate hardware Model with very fast Chipset (not available in 19'' Versions):



You can detect the Web-IF Firmware in the STATUS Page. This Hardware shows a Firmware Version with 2.xy and an 'A' which reflects to the Chipset.

Some addons/changes in this Version: System settings:



-TS and RTSP streams can be used with the user/password option (same as web-IF-login)

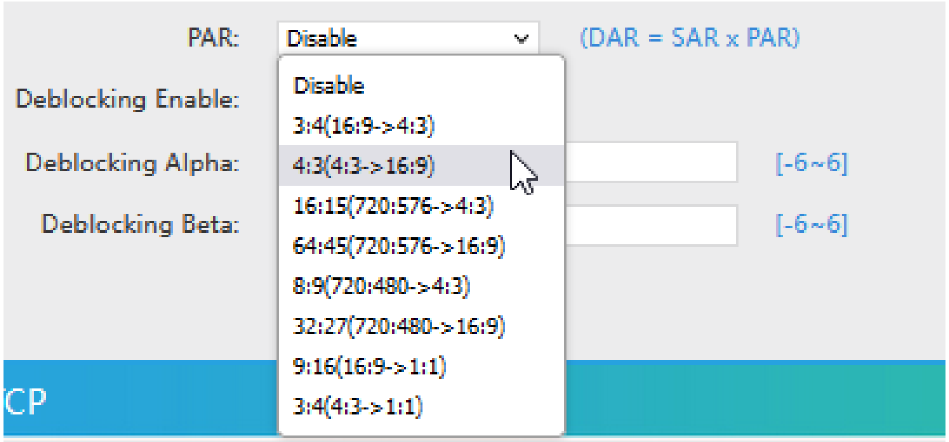
-The web-login user/pw can be disabled

-Telnet access can be switched ON or OFF

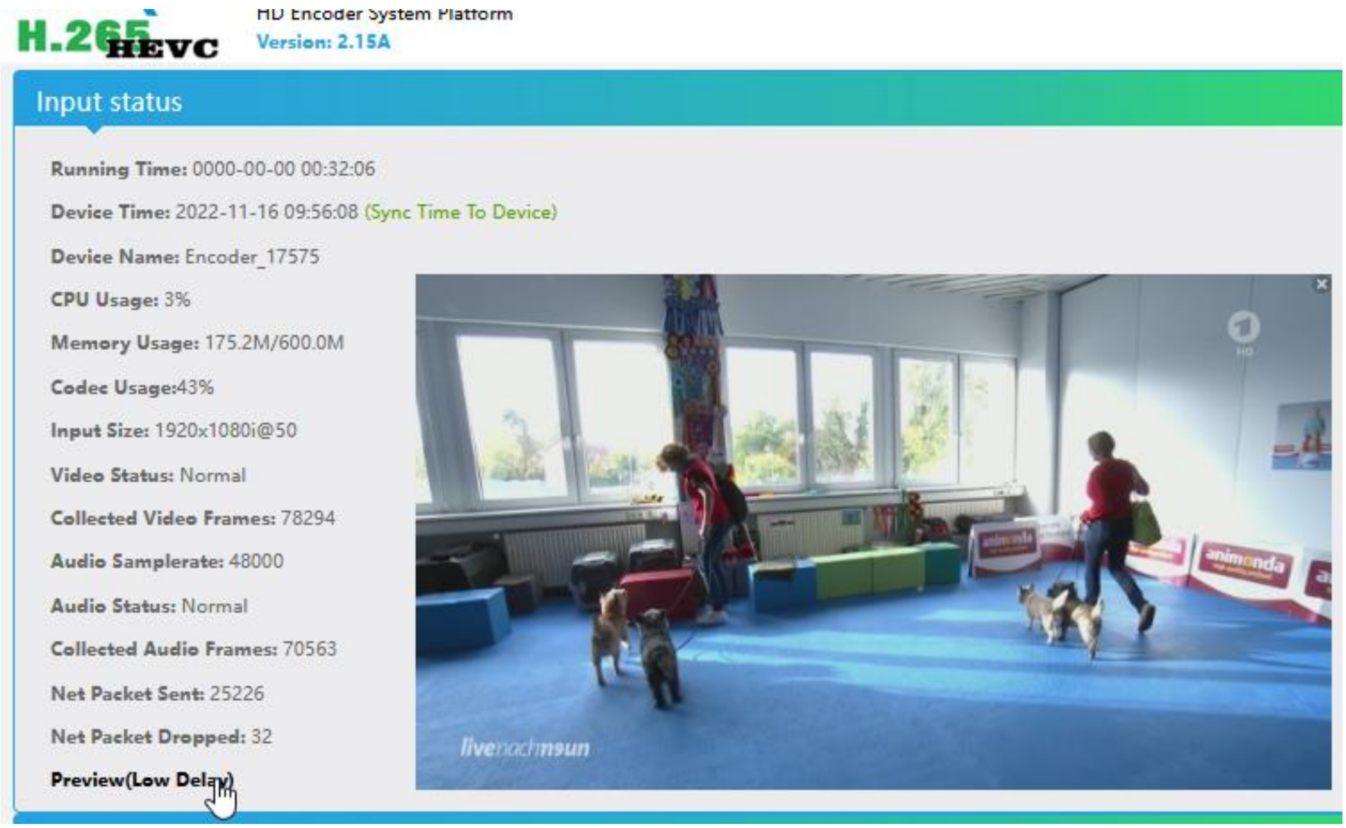
-RTSP selection as a default stream from which encoder part

- SAP enable/disable

PAR= Picture Adjustment's: Format transforming's as well as De-blocking feature:



and an Input-Preview Window:



Version 5.23 has some improvements in GOP encoding processes.

Technical Specification SDE-16265:

Function	h.265 (HEVC compatible) and h.264 (MPEG4 compatible) Encoder and IP Streamer
INPUT	HD-SDI / SD-SDI (BNC type) input and loop through output, Level detection = Auto V.5.17 on
Resolution	1080p, 1080i, 720p and below
Video encoder	h.265 (HEVC) or h.264 (AVC) or MJPEG
Audio encoder	AAC, AAC++, MP3, MPEG1Layer2, AC-3 stereo compatible
Audio Bit-rate:	Bit-rate: 32k/48k/96k/128k/160k/192k, Data-rate: 64 kbps-384 kbps
SYSTEM	
Data interface	RJ45, 1000BaseT Ethernet interface, management by web browser
Protocol	HTTP, RTSP, RTMPs, UDP/RTP, FLV, HLS ; unicast/multicast, SRT P2P
Data Rate	32 kbps – 32 Mbps
Encoding bitrate process	CBR or VBR
SMPTE 425	Support Level A & B
GOP Structure	IBBP
ONVIF 2.x	Supported by RTSP: G711A
Picture adjust	De-interlacing, Noise reduction, Sharpening
OSD	4 Logo and Text Insertion as transparent overlays possible
Power supply	2x redundant internal switching PSU‘s 110…240VAC 47-64Hz to 12V DC
Dimensions	19“ 3RU: 483 (L) x 300 (W) x133 (H) mm
Weight	6,5 kg
Consumption	90W

Finally: To get more information about the deeper details of the encoder settings and configuration issues, please check some of the combined PDF – Manuals from our website or write us.

Contact:

info@blankom.de www.blankom.de

Annex A: A Guide to Encoding and Low Latency Streaming:

From Video Encoding Basics to Optimizing Streaming Workflows

Video streaming over the internet is gaining importance in many industries, including broadcast, enterprise, and government. It has become popular for a number of reasons, particularly because live video is a great way to contribute content and engage with consumers, employees, and the community. For broadcast engineers video streaming over the internet is a cost-effective and flexible alternative to satellite services. For AV professionals, video streaming, if correctly implemented, can be an efficient and flexible means of communicating across an organization. Flexibility is not only important for content creation and keeping up with demand, but also for scalability and business continuity.

Video streaming begins with video encoding. For content creators, video encoding can be the most important part of a workflow, which is why it is so important to have a solid grasp of the basics of video encoding before embarking with video streaming.

This guide will explore the principle concepts of video encoding and streaming, including compression, codecs, latency, and network transport considerations when streaming from encoders.

The best way to identify your encoding priorities is to first understand the use case and end goal for your live video stream – what is it needed for, for who, and what will be the measure of its success? By establishing these priorities, one can review the four factors that make up a successful live video streaming experience:

- Quality
- Bandwidth
- Security and Reliability
- Latency

AN INTRODUCTION TO VIDEO ENCODING

What is Video Encoding?

Video encoding is the process of compressing raw video for transport over IP networks such as office LANs and the internet. As IP networks have limited bandwidth, the encoder needs to be able to compress the content accordingly. There are two types of video encoding: file-based and live, and it's important to make the distinction between them.

When working with video files, encoders are used to compress and reduce the size of video content so that it can take up less storage space and be easier to transfer from one part of a video production workflow to another. Since the video files are not live, latency is usually not a key concern.

Live video encoding is the process of compressing real-time video and audio content prior to streaming. Compression significantly reduces the bandwidth required, making it possible for real-time video to be transmitted across constrained networks while maintaining picture quality at levels suitable for viewing. However, depending on the type of encoder used, compressing live video can also add latency which if too great can negatively impact the overall quality of experience.

Decoding and Transcoding: A Brief Overview

Video decoding is essentially the opposite of encoding. It is the process of decoding or uncompressing encoded video. A decoder can output uncompressed video through SDI for further video processing or over HDMI for displaying directly on a screen.
Decoders can also extract embedded audio tracks for sound production. Embedded metadata can be passed on by the decoder to other production components for information on video formatting, time codes, subtitles, and closed captioning.

Synchronizing Feeds Some decoders, support multiple incoming streams and can resync them based on timecode prior to decoding to SDI. This is especially useful for live broadcasts with multiple Camera angles that share an audio source.

For live video, it is imperative that video decoders add as little latency as possible in order to minimize the impact on production and provide a broadcast quality experience.

Video transcoding is the process of converting an already encoded stream from one format to another, or from one size to another. Most transcoders use a two-step process of decoding and re-encoding. Video transcoding is commonly used for enabling OTT (over the top) internet streaming services with a high quality source or mezzanine video transcoded into a cascade of different bitrates and resolutions. These multiple video transcodes or profiles are needed for ABR (adaptive bitrate) streaming which adapts picture quality in real-time based on available bandwidth. This enables a single video source to be delivered to different viewing devices including connected televisions, computers, and smartphones.

Video Encoders

There are two types of video encoders - software and hardware-based.

Software encoders can be installed on standard off the shelf hardware or as virtual machines (VM) in data centers and cloud platforms. Although software can be a great option for encoding file-based video content, depending on the computer hardware they run on, they don't always offer ultra-low latency levels like dedicated hardware encoders and therefore are not always suitable for live broadcast contribution applications.

Hardware encoders are turnkey devices with dedicated processing power for low latency encoding of video streams. Whereas software encoders have to share CPU and other resources, hardware encoders can use purpose-designed micro-processing chips and can therefore encode and stream live video with very little latency.
Hardware video encoders are used by a wide range of organizations for delivering pristine quality, low latency video for many different applications including:

- Broadcast – for backhaul, bi-directional interviews, return feeds, and remote production (REMI)
- Enterprise – for internet streaming of all-hands meetings, product training, and employee briefs as IPTV, and digital signage
- Defense – for mission critical Intelligence, Surveillance and Reconnaissance (ISR) applications

VIDEO CODECS AND COMPRESSION

What is a Video Codec?

The term codec is a portmanteau of the words ENCoding (coding) and DECoding. It describes a process for compressing and decompressing data as files or real-time streams. For engineers, a codec usually refers to the compression format used by a video encoder, decoder, or transcoder.
Codecs for live video, mainly H.264/AVC or H.265/HEVC, can reduce raw content data by as much as 1,000 times, saving much-needed bandwidth and enabling real-time video streams or files to be easily transmitted across constrained networks and to end-user devices. For example, a typical uncompressed HD stream is about 1.5 Gbps and is compressed to around 5 Mbps for live broadcast television.

Bitrates A bitrate is the number of video data bits that are processed within a unit of time and is commonly measured in bits per second, often abbreviated as bps. Generally, higher bitrates result in higher image quality, but newer and more complex codecs, like HEVC, often deliver better video quality at lower bitrates than older codecs.

VIDEO COMPRESSION TECHNIQUES

The compression techniques used by different codecs can fall into two main categories: lossy and lossless.

Most codecs use “lossy” compression methods which, at a high level, means that when a video is compressed, some redundant spatial and temporal information is lost. “Lossless” compression is used when the goal is to reduce file and stream sizes by only a slight amount in order to keep picture quality identical to the original source.

Lossless compression formats such as JPEG-XS can reduce bit-rates up to 10 times with no loss of picture information and are well-suited for supporting SMPTE 2110 workflows within production studios.

The reduction needed for compression methods can be divided into two main categories: intra-frame and inter-frame. Intra-frame compression or spatial reduction, reduces the size of each individual frame within a video file or stream. Examples of intra-frame formats are JPEG-2000, used for high bitrate broadcast contribution and video archiving, and JPEG-XS used for SMPTE 2110 workflows.

Inter-frame compression or temporal reduction works by grouping multiple frames within a group of pictures or GOP, and encoding only the pixels that change between consecutive frames, based on an initial reference or key frame, known as the I-frame. AVC / H.264 and HEVC / H.265 are the most common inter-frame compression formats as they offer significant bitrate reductions suitable for streaming SD, HD and 4K video content.

Framing Options

Choosing the right combination and number of I, P, and B-frames is key to optimizing video quality. The main reference comes from the I-frames, which contain the most amount of data. P-frames only contain the differences between it and the previous I-frame, and B-frames contain both forward and backward changes resulting in even more efficient compression.

Here are some examples of framing options available with high end encoders:

I: I-frames only (highest quality, least amount of bandwidth efficiency) IP: I and P-frames only (high picture

quality, efficient compression) IBP: I, B and P-frames

IBBP: I, BB (two B-frames in sequence) and P-frames

IBBBP: I, BBB (three B-frames in sequence) and P-frames

IBBBBP: I, BBBBB (four B-frames in sequence) and P-frames (highest latency; highest bitrate efficiency)

MPEG-2 MPEG-2 has been around since the 1990s and introduced the world to digital television and DVDs. It is slowly being phased out, but used in many legacy applications and terrestrial broadcast systems.	**JPEG2000** JPEG2000 is a codec used for digital cinema, medical imaging, geospatial data, and document archiving.
High quality Not efficient	Intra-frame encoding for high image quality Requires a lot of bandwidth and storage space
H.264 / AVC (Advanced Video Coding) H.264 makes up the majority of multimedia traffic; it is used for high quality streaming and HD television. However, as the demand for 4K continues, it is being replaced with HEVC.	**H.265 / HEVC (High Efficiency Video Coding)** The successor to H.264, HEVC or H.265, is fast becoming ubiquitous thanks to the proliferation of 4K content.
Fast encoding speed Efficient for HD video Not efficient enough for 4K UHD	Highly efficient – can deliver same quality as H.264 at half the bitrate Excellent quality – ideal for 4K UHD video Requires significant processing power – not always suitable for software encoders
VP9 VP9 is a royalty-free and open source codec developed by Google and used by YouTube, but with few features, is rarely used in commercial or broadcast applications.	**JPEG-XS** JPEG-XS is a lightweight “lossless” compression standard which is useful for 4K and 8K contribution workflows over existing 3G SDI and 12G SDI or hybrid SDI/IP networks including SMPTE-2110.
Open source Few benefits compared to codecs like HEVC Limited availability of real-time VP9 encoders	Ideal for sending 4K video over a 10GB network to devices that aren’t equipped for 12GB Limited configuration settings
AV1 AV1 is a newer open and royalty-free standard. However, it is still very much a work in progress.	**VVC (Versatile Video Coding)** VVC is a next-generation compression standard, in its early development phase. First hardware implementations are slated for later in 2021.
Open source and evolving Requires significantly more computing power than HEVC A lack of details has caused hesitation on its adoption	More efficient that HEVC – it is focused on achieving 30% better compression efficiency Still in development and will not be tested in real-world implementation for a while

THE MUST-HAVE CODECS – HEVC AND H.264

In most use cases, broadcast engineers and AV professionals need to be able to support both H.264 and HEVC depending on the source, video format, and devices they are streaming to. H.264 is often required for SD and HD content streamed to digital broadcast systems, displays, and set top boxes, while HEVC is rapidly being deployed by organizations with a focus on live video workflows and 4K content.

Today's live video encoders need to support either H.264, HEVC, or both, as they are the most widely used and fit for purpose codecs for streaming over IP networks. Encoding in H.264 or HEVC also ensures interoperability with all types of decoders, viewing devices, and web browsers.

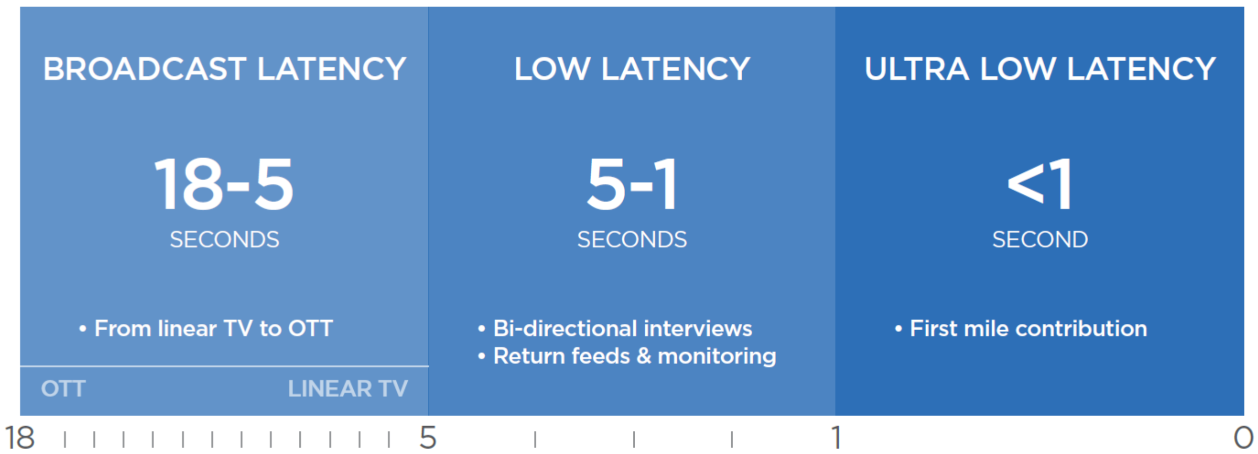
LATENCY

What is Video Latency?

Within the context of video streaming, latency is the amount of time it takes for video to travel from its initial source to its final destination. The complete journey from camera to screen is often referred to as glass-to-glass or end-to-end latency. Encoder latency describes how long it takes for a raw video stream to be processed and encoded prior to streaming, while network latency is the amount of delay on a given IP network. As multiple encoders, transcoders, decoders, and other video processing components may be sharing more than one type of network, these all add up to the total end-to-end latency.

Low end-to-end latency, at under 5 seconds, is critical for broadcasting live events, whereas OTT services are typically delivered with up to 15 seconds of latency. However, for specific tasks such as broadcast contribution, bidirectional interviews, or reconnaissance missions, latency levels need to be much lower or well under 1 second to ensure minimal impact on overall end to end latency.

LIVE VIDEO STREAMING LATENCY



Live video streaming latency

Sources of Latency

There are several factors which contribute to end-to-end latency including the complexity of the content delivery chain and the number of video processing steps involved. While individually these delays might be minimal, cumulatively, they can add a disruptive delay that compromises the viewing experience.

Key contributors to video latency include:

• Individual components in the streaming workflow

Each component in a streaming workflow, including cameras, encoders, decoders and display devices, create delays which contribute to latency in varying degrees. Latency continues to rise with each additional component used to process the video content prior to delivery.

• Network type and speed

The network chosen to transmit a video (ex: public internet, satellite link, or MPLS network) impacts both latency and quality. Network speed is generally defined by throughput or how many megabits or gigabits can it handle in the course of a second and also by the distance traveled.

• Streaming protocols and output formats

The choice of video protocol also impacts video latency. A transport protocol like SRT is low latency while an ABR protocol like HLS is high latency. In addition, the type of error correction used by the selected protocol to counter packet loss and jitter can also add to latency.

Ultra-Low Latency

Broadcast engineers work to keep overall latency as low as possible from the start – keeping the delay from the camera to the production studio at less than 1 second (or ideally under 300 milliseconds) referred to as ultra-low latency.

LIVE LATENCY

For live video, latency should be kept as low as possible. This helps to ensure an optimal viewing experience across a variety of content; low latency prevents news and sports broadcasts from being "spoiled" by social media, and eliminates any awkward pauses that can otherwise arise in bi-directional interviews.

However, a degree of latency can be useful, even for live applications. Introducing delay between broadcast production and playout can also facilitate live subtitling, closed captioning, and prevent obscenities from airing.

When Milliseconds Matter

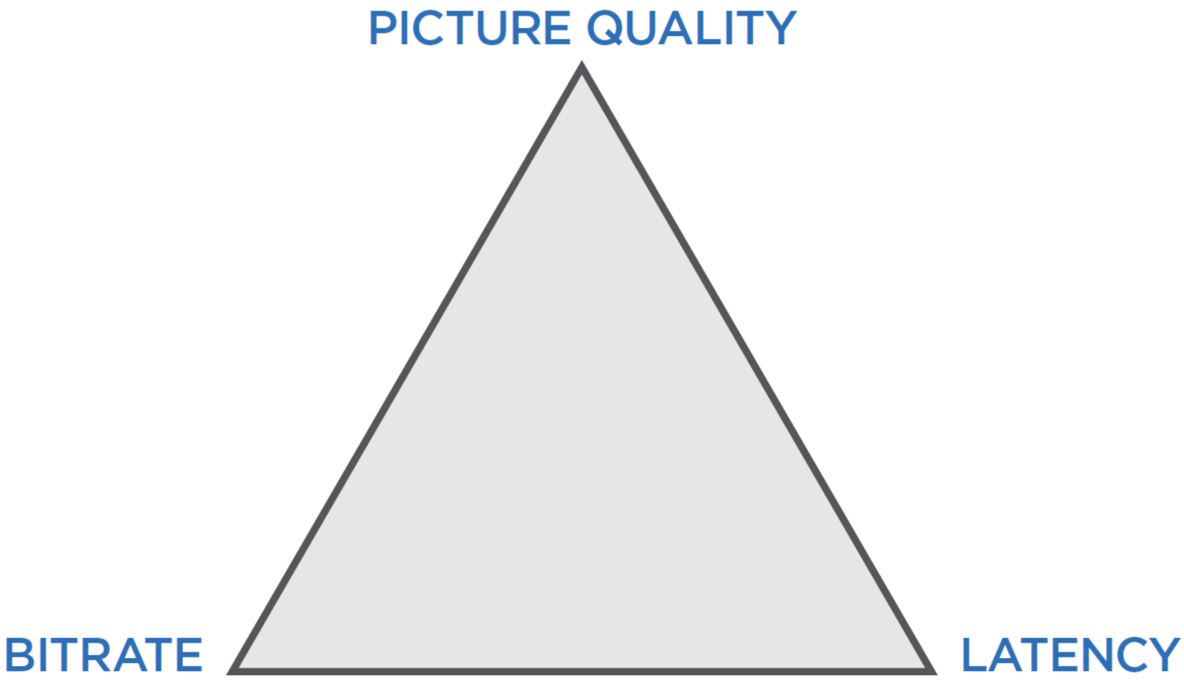
While video latency presents a serious annoyance to sports fans, low latency is also a critical requirement for surveillance missions, online auctions, and gaming applications.

How to Reduce Latency

There are a few ways to minimize video latency without having to overly compromise on picture quality.

The first is to choose a hardware encoder and decoder combo engineered to keep latency as low as possible, even when using a standard internet connection. The latest generation of video encoders and video decoders are able to maintain low latency (under 50ms in some cases) and have enough processing power to apply the HEVC codec to compress live video to extremely low bitrates (down to under 3 Mbps for HD content) all while maintaining high picture quality.

Balancing Latency with Picture Quality and Bandwidth Availability



Balancing Bitrate, Latency, and Picture Quality

These three factors need to be taken into consideration when encoding and streaming live content.

Any video encoder used for broadcast quality live video streaming should allow users to change bitrate, picture quality, and latency settings. Ultimately, the individual targeted use case will determine the best balance within this triangle of video encoding and streaming considerations.

Latency is not the only consideration in a video streaming workflow; video quality and bandwidth are also very important across each application. The three operate in a balance: when the latency is reduced, either the picture quality will be reduced, or the bitrate will increase.

Network Adaptive Encoding (NAE)

Some encoders feature Network Adaptive Encoding. NAE enables encoders to automatically adjust compression levels based on real-time network bandwidth information.

For applications where latency is critical such as video surveillance and ISR, picture quality can often be exchanged in favor of minimizing latency. However, for use cases where pristine broadcast quality video matters, latency can be increased slightly in order to support advanced video processing and error correction. By delivering the optimal combination of bandwidth efficiency, high picture quality, and low latency, viewers can enjoy a great live experience over any network.

Another important factor in keeping latency low is between the encoder and decoder – video transport, which is the subject of the next chapter.

VIDEO STREAMING

While streaming takes place after the video encoding process, it is important to consider the different applications and the best transport protocol for the job.

What Is a Transport Protocol and when is it Used?

A transport protocol is a communication protocol responsible for establishing a connection between two or more devices and delivering data across a network. Transport protocols provide data delivery guarantees that are essential for file transfers and mission-critical applications. Especially important for video streaming, different transport protocols may support a range of optional capabilities including error recovery, flow control, and support for re-transmission.

Important Factors in Choosing a Transport Protocol

- Latency: Different protocols will introduce different amounts of latency to your workflow; live applications will require a transport protocol with as low latency as possible.
- Security: Regardless if the content is for broadcast, government, or enterprise purposes, security is paramount. Video streams being sent over the public internet will need a transport protocol that enables them to be encrypted.
- Reliability: Networks like the public internet can be unreliable, which can result in the loss of data in transport ("packet loss"). This packet loss can degrade the quality of the video, unless the transport protocol in question has a means of error correction, recovering these lost packets.
- Flexibility: Some video streams need to be sent across a room; others across an ocean. Depending on how far a video must travel, it may have to cross multiple firewalls, necessitating a transport protocol that can make firewall Traversal easy.

Error Correction: FEC vs ARQ

FEC (forward error correction) and ARQ (automatic repeat request) are the most common methods of packet loss recovery in video streaming. In FEC, the encoder sends a duplicate of each packet, in case one is lost in transport. With ARQ, the encoder labels each packet, allowing the decoder to request a packet be re-sent should it be lost in transport.

TRANSPORT LAYER PROTOCOLS

The two fundamental transport protocols used by the internet for sending and receiving data include UDP and TCP. UDP is typically used for live streams while TCP is used for file-based content. Other transport protocols such as RTMP or SRT are higher layer protocols that are based on UDP and TCP.

TCP (Transmission Control Protocol,) is the most commonly used transport protocol on the internet. TCP is considered to be very reliable as packets are tracked in such a way that no data is lost or corrupted in transit. In spite of its apparent reliability, TCP is unsuitable for live video; packets are often delayed due to error recovery and they are not always received in the right order.

UDP (User Datagram Protocol,) works similarly to TCP, but is far faster, as it uses different error recovery methods. When video streaming with UDP, “datagrams” (essentially packets) are received by the decoder in the order they were sent. However, unless the application layer protocol being used has its own error recovery method, there is no recourse if packets are missing. UDP is very fast and therefore frequently used for time-sensitive applications such as online gaming or live broadcasts where perceived latency is more critical than packet loss.

COMMON TRANSPORT LAYER PROTOCOLS

RTMP (Real-Time Messaging Protocol)

RTMP is a legacy, but still commonly used protocol for live streaming within production workflows. Based on TCP, RTMP is a continuous streaming technology with packet loss recovery, though at the cost of increased latency. Because RTMP does not support HEVC, it is being phased out by broadcasters and CDNs.

SRT (Secure Reliable Transport)

SRT is an open-source streaming protocol that enables AES 128/256 bit encryption to keep streams secure. It utilizes ARQ packet recovery to maintain high quality over unreliable networks without compromising latency, but will also support FEC packet recovery for those who prefer the latter. In 2019, the protocol was awarded an Emmy for Engineering and Technology. Usually not in use in Headend-Environment’s.

RTP (Real-Time Transport Protocol)

RTP is an internet protocol for real-time transmission of multimedia data in unicast or multicast mode. RTP runs over UDP for low latency and though it does not include packet-loss recovery it has mechanisms to compensate for any minor loss of data when used in conjunction with the RTP Control Protocol (RTCP) for monitoring quality of service.

RTSP (Real-Time Streaming Protocol)

RTSP allows viewers to remotely pause, play, and stop video streams via the internet without the need for local downloads. This application layer protocol was most notably used by Real Networks RealPlayer and is still being applied for various uses including for remote camera streams, online education and internet radio. RTSP requires a dedicated server for streaming and does not support content encryption or the retransmission of lost packets as it relies on the RTP protocol in conjunction with RTCP for media stream delivery.

Delivery Protocols

HLS and MPEG-DASH are two commonly used streaming protocols for video content delivery. Both of these protocols support adaptive bitrate streaming (ABR) and are therefore ideal for delivering produced content to viewing devices over an internet connection. However their high level of latency make them unsuitable for video contribution.

Proprietary Protocols

There are a number of proprietary protocols based on TCP and UDP, which are designed for video delivery, however they do require a license which can add costs and also prevent interoperability with third-party vendor equipment.

CHECKLIST: CHOOSING A VIDEO ENCODER

There are several important factors involved in choosing a video encoder. For those who are new to video streaming, it can feel a little overwhelming, while individuals who are accustomed to certain systems may be nervous about switching components.
Regardless of the workflow, there are important considerations to bear in mind when selecting a video encoder, as we outline in our checklist.

Use Case

The single most important question to ask when looking for a video encoder. By establishing the purpose of the streamed video content, you can prioritize the elements that a video encoder can bring to your workflow.

Form Factor

The form factor of a video encoder is very important to a streaming setup, whether you have the space for a full hardware video encoder, or simply a blade. Some situations require ruggedized video encoders for harsh environments, especially when operating outdoors.

Codecs

It is important that the video encoder you are using is capable of supporting the video codec you want to use, in most cases HEVC or H.264. Some video encoders support both of these codecs, which is an excellent way of future- proofing your streaming setup.

Transport Protocol

Choose a video encoder that supports the video transport protocols best suited to your streaming application; for streaming encrypted video at low latency for broadcast contribution or live event production, consider an encoder that supports SRT.

Latency

To keep latency low in a video streaming workflow, you need to start from the beginning. If the video encoder is adding latency, there won't be a way to "catch up" on that delay later in the streaming process. Therefore, choosing a video encoder with low latency is crucial for minimizing total delay.

Interoperability

A video encoder is part of a larger workflow or setup; it is important that it fits with the other pieces. Ensure that your video encoder is compatible with the other elements of your workflow including your camera and decoder.

Ease of Use and Support Availability

This is more a function of who is operating the video encoder. While many broadcast engineers will be familiar with a variety of encoders and hardware, IT managers at corporate offices or volunteers at a religious or community organization may need a more user-friendly device with support available to them.

Reliability

Depending on the situation, some video encoders may have to contend with unpredictable networks. In order to ensure the reliability of the video stream, some video encoders have options like adaptive bitrate encoding to ensure that the best quality video possible is always available.

Quality

The video quality needed will greatly influence the kind of encoder required. There are video encoders that can stream in HD, and 4K UHD – although not all workflows prioritize that level of quality.

Security

Keeping video streams secure helps protect intellectual property and prevents unauthorized access to videos. Choosing a video encoder with security options like AES 128/265-bit encryption will ensure that streams are safe from the start.