Setup of Output Streams

This article will show how to setup your Output Streams for different purposes.

In this article:

---

1. Introduction

You are able to setup 15 different Stream Configurations for any Source, being Channel, Inputs and Director View. As a default, your Video Format is copied from the Source, once the Stream starts, but you can transcode into any Resolution.

Once you pick your Protocol, the other Fields for Video/Audio Codec and URL will change according to your selection, meaning: Not every Video/Audio Codec will be available for every Stream Protocol.

INFO: NVIDIA Encoder do NOT support Interlaced Formats anymore. Pick Intel QuickSync instead.

Your URL will show a suggestion of the required URL Format. Click on “URL Info” to show more examples, e.g. how to integrate Username/Password. The Preview URL will not be used by PLAYDECK, it’s just for your convenience to open the Internet Browser upon clicking “Preview”.

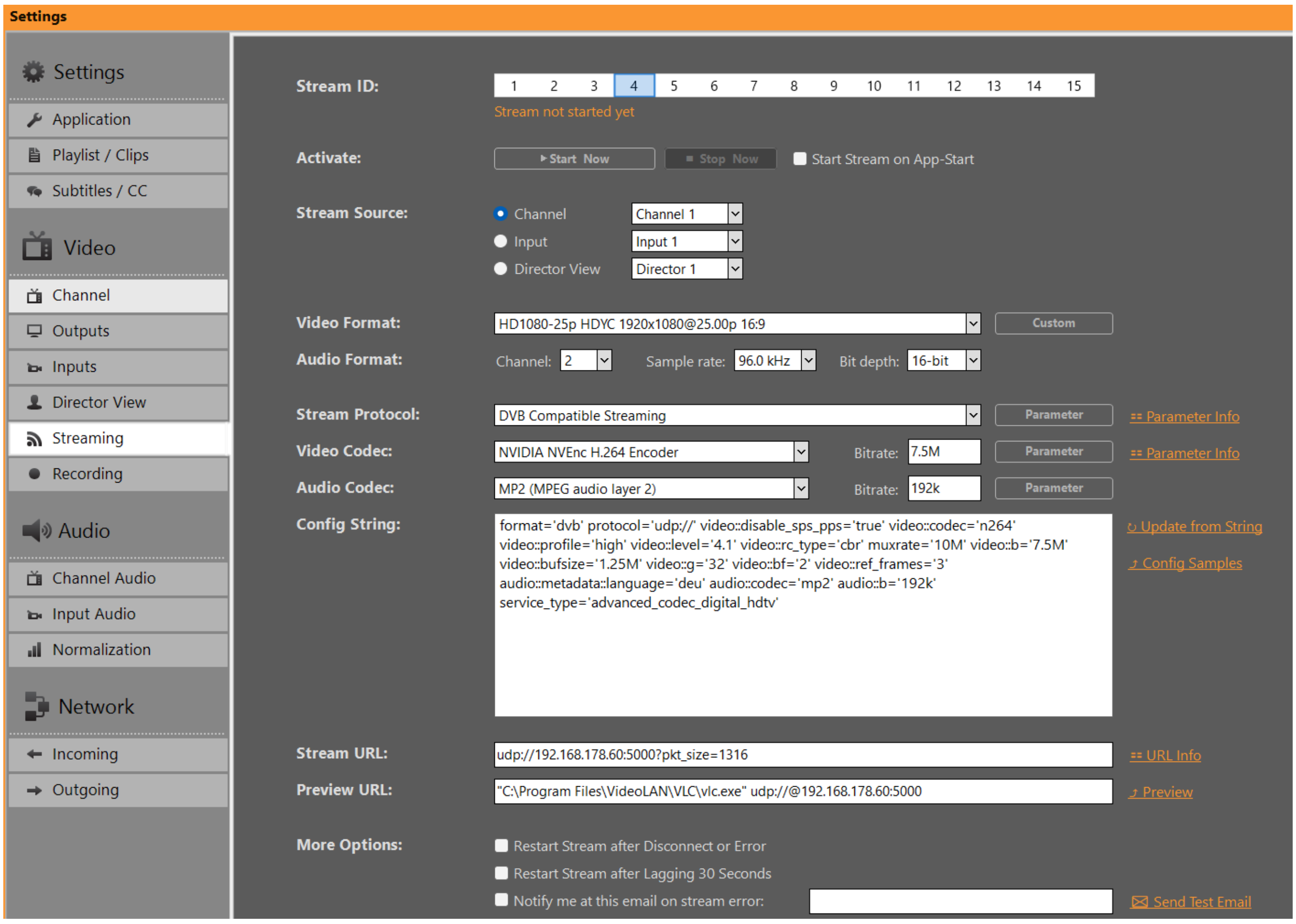
---

2. Using Parameter and Config

Further Parameter can be set to Protocol and Codecs, which are specific for your selection. These can be used to finetune your Settings, e.g. make your Stream CBR or add Metadata.

Clicking on “Parameter” will open a Editor, which will only show the possible Settings for your selection. It will highlight any Value that differs from the Default Value.

The “Config String” is a representation of all your Protocol/Codec selection as well as their respective Parameter. The Config String works both ways: You can also copy/paste a Config String into the Text Field, then click “Update from String” to apply the Values upwards. This allows you to simply copy Settings between Stream ID’s or share with other User.



Settings

- Settings
  - Application
  - Playlist / Clips
  - Subtitles / CC
- Video
  - Channel
  - Outputs
  - Inputs
  - Director View
  - Streaming
  - Recording
- Audio
  - Channel Audio
  - Input Audio
  - Normalization
- Network
  - Incoming
  - Outgoing

Stream ID: 1 2 3 [✓ 4] 5 6 7 8 9 10 11 12 13 14 15
Stream not started yet

Activate: Start Now | Stop Now | Start Stream on App-Start

Stream Source: [✓ Channel] Channel 1 / Input: Input 1 / Director View: Director 1

Video Format: HD1080-25p HDYC 1920x1080@25.00p 16:9 | Custom

Audio Format: Channel: 2 | Sample rate: 96.0 kHz | Bit depth: 16-bit

Stream Protocol: DVB Compatible Streaming | Parameter | Parameter Info

Video Codec: NVIDIA NVEnc H.264 Encoder | Bitrate: 7.5M | Parameter | Parameter Info

Audio Codec: MP2 (MPEG audio layer 2) | Bitrate: 192k | Parameter

Config String:

```
format='dvb' protocol='udp://' video::disable_sps_pps='true' video::codec='n264'
video::profile='high' video::level='4.1' video::rc_type='cbr' muxrate='10M' video::b='7.5M'
video::bufsize='1.25M' video::g='32' video::bf='2' video::ref_frames='3'
audio::metadata::language='deu' audio::codec='mp2' audio::b='192k'
service_type='advanced_codec_digital_hdtv'
```

Update from String | Config Samples

Stream URL: udp://192.168.178.60:5000?pkt_size=1316 | URL Info

Preview URL: "C:\Program Files\VideoLAN\VLC\vlc.exe" udp://@192.168.178.60:5000 | Preview

More Options:
- Restart Stream after Disconnect or Error
- Restart Stream after Lagging 30 Seconds
- Notify me at this email on stream error: | Send Test Email

3. Config Samples

Here is a quick selection of recommended Configurations for different purposes. Copy and Paste to PLAYDECK and click “Update from String”.

DVB Stream with CBR and GOP B-frames. No PCR or PTS-DTS or Buffer underrun Error

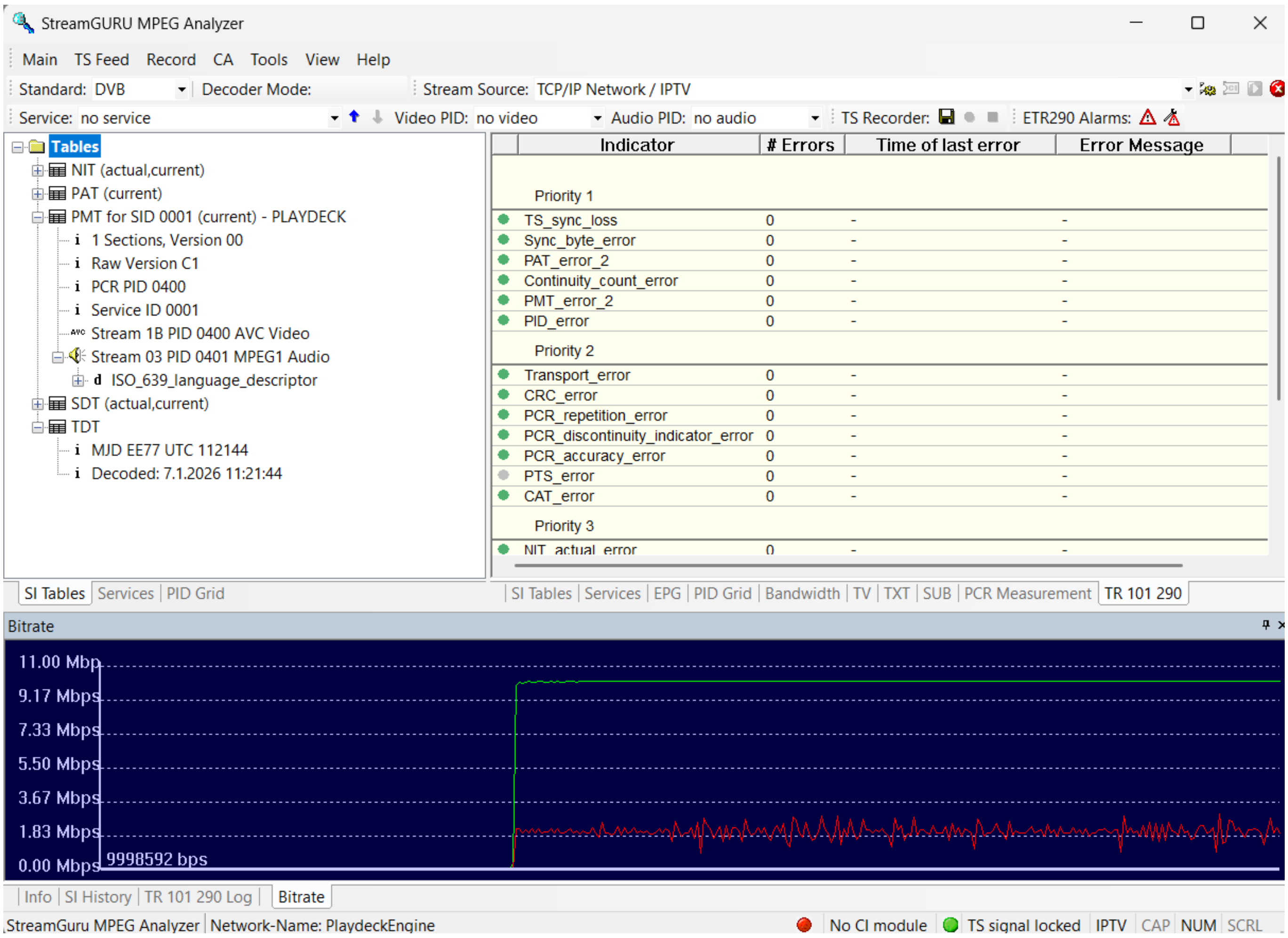
```
format='dvb' protocol='udp://' mpegts_start_pid='60' video::streamid='80'
video::disable_sps_pps='true' video::codec='n264' video::profile='high'
video::level='4.1' video::rc_type='cbr' muxrate='10M' video::b='7.5M'
video::bufsize='1.25M' video::g='32' video::bf='2' video::ref_frames='3'
audio::streamid='81' audio::metadata::language='deu' audio::codec='mp2'
audio::b='192k' pmt_start_pid='100' mpegts_flags='-system_b' rc_buffers='4'
pcr_period='35' pat_period='0.1' sdt_period='0.5' tdt_period='10'
start_pcr='0' service_name='TEST' service_provider='PD' service_id='1'
original_network_id='1' transport_stream_id='201'
service_type='advanced_codec_digital_hdtv'
```

Or in Short Form:

```
format='dvb' protocol='udp://' video::disable_sps_pps='true'
video::codec='n264' video::profile='high' video::level='4.1'
video::rc_type='cbr' muxrate='10M' video::b='7.5M' video::bufsize='1.25M'
video::g='32' video::bf='2' video::ref_frames='3'
audio::metadata::language='deu' audio::codec='mp2' audio::b='192k'
service_type='advanced_codec_digital_hdtv'
```

INFO: Every DVB Receiver is different und you might need different Parameters. Some Receiver will not allow ANY deviancy from official guidelines, and will show BLACK. The most reliable solution to encode DVB signals are Hardware Encoder, e.g. Haivision.

Best practice to test DVB Streams for PCR Error or other, is using one of these Tools:
– DVBControl DVBAnalyzer (costs ~2000 EUR, but is most reliable and detects PTS-DTS Error)
– StreamGURU (costs ~300 EUR, can detect any ETR Error and shows Bitrate Graphic)
– TSAnalyzer (free, but can ingest Stream URL, needs TS File on HD)
– Others worth mentioning: DVBInspector, Elecard Stream Analyzer, StreamXpert



Standard: DVB | Decoder Mode: | Stream Source: TCP/IP Network / IPTV
Service: no service | Video PID: no video | Audio PID: no audio | TS Recorder: | ETR290 Alarms:

- Tables
  - NIT (actual,current)
  - PAT (current)
  - PMT for SID 0001 (current) - PLAYDECK
    - 1 Sections, Version 00
    - Raw Version C1
    - PCR PID 0400
    - Service ID 0001
    - Stream 1B PID 0400 AVC Video
    - Stream 03 PID 0401 MPEG1 Audio
      - ISO_639_language_descriptor
  - SDT (actual,current)
  - TDT
    - MJD EE77 UTC 112144
    - Decoded: 7.1.2026 11:21:44

	Indicator	# Errors	Time of last error	Error Message
	Priority 1			
●	TS_sync_loss	0	-	-
●	Sync_byte_error	0	-	-
●	PAT_error_2	0	-	-
●	Continuity_count_error	0	-	-
●	PMT_error_2	0	-	-
●	PID_error	0	-	-
	Priority 2			
●	Transport_error	0	-	-
●	CRC_error	0	-	-
●	PCR_repetition_error	0	-	-
●	PCR_discontinuity_indicator_error	0	-	-
●	PCR_accuracy_error	0	-	-
●	PTS_error	0	-	-
●	CAT_error	0	-	-
	Priority 3			
●	NIT_actual_error	0	-	-

StreamGuru MPEG Analyzer | Network-Name: PlaydeckEngine | No CI module | TS signal locked | IPTV | CAP | NUM | SCRL

RTMP Stream in CBR for 60 FPS with 2 second-keyframe

The next Sample is for YouTube. Use the YouTube Analytic and Stream Health

Tools to test your Stream Satbility:

```
format='flv' protocol='rtmp://' video::codec='n264' video::b='9000k'
video::maxrate='9000k' video::bufsize='9000k' video::g='120'
video::preset='veryfast' audio::codec='aac' audio::b='128k' audio::ar='44100'
```

The next Sample is for Twitch. Use Twitch Inspector to check your Stream Stability and add “?bandwidthtest=true” to your Stream URL to prevent going Live.

```
format='flv' protocol='rtmp://' merge_tracks='true' video::codec='n264'
video::b='5800k' video::maxrate='5800k' video::bufsize='5800k' video::g='120'
video::preset='veryfast' audio::codec='aac' audio::b='128k'
video::tune='zerolatency' audio::codec='aac' audio::ar='44100'
audio::b='128k'
```

---

4. AAC High Quality Audio Streams

The default AAC Encoder for Stream will be very conservative, meaning: Use as little CPU as possible and leave maximum room for video bandwidth. If your project is to stream music channel, this might not be satisfying enough.

The first thing you might want to consider is dropping video res to 720p30, so you have more bandwidth for audio. Especially if you run multiple streams. Always use a Hardware Encoder for video like H264 NVidia NVENC.

For audio encoding, using AAC is your only true choice for high quality audio. AAC is available in most stream formats (RTMP, SRT, UDP) and has much less compression compared to MP3.

Next, you always want to go with 48KHz and 32 bits as a default, since AAC can use full dynamic.

Here is a sample config, which we will explain in more detail:

```
format='mpegts' protocol='udp://' video::codec='n264' video::b='4M'
audio::codec='aac' audio::b='256K' audio::profile='aac_low'
audio::stereo_mode='ms_force' audio::minrate='200K' audio::maxrate='320K'
audio::bufsize='640K' audio::aac_coder='0'
```

Since we saved some bandwidth on video, we can raise our Bitrate to 256K. We set a Minrate of 200K, so silent passages will not sound rough. And we set a Maxrate of 320K to allow for peaks. The Buffer Size of 640K represents approx 2,5s Buffer. If you only run one music channel, you can even go higher on the bitrates.

Now to some very important settings:
The AAC Profile must be “aac_low”, which is AAC-LC and has the highest quality and supports higher bitrates. We force Stereo Audio Encoding with “stereo_mode=ms_force”. We set the AAC Coder to “0”, which means “double encoding run”. Reduce this to “1” (single run), if you get CPU problems.

Avoid “2”, which means “single fast run”.

One more thing to mention:
Use source material (clips), which are 48 KHz, 32 bit and high bitrate of at least 320K VBR. This is important to not introduce audio artifacts, which will be amplified by the AAC Encoder later on. If you only run one music channel, you can try lossless audio (PCM) and ~2000K Bitrate.

---

5. Color Space and Chroma subsampling and Interlace

The Support for certain Color Spaces, e.g. 4:2:2 for DVB, or Interlaced Encoding, depend very much on the Intel CPU or NVidia GPU you are using.

Here is a page for NVidia NVenc and 422 Support and here for Intel Quick Sync and 422 Support.

In basic terms:

1. NVidia Support for Interlaced Encoding:
– Pascal (GTX 10-Serie): Last architecture with interlace support
– Turing / Ampere / Ada (RTX 20/30/40): Hardware support for interlaced encoding was removed
– Blackwell (RTX 50-Serie): Interlaced encoding for H.264 (8-bit and 10-bit) has been re-introduced

2. NVidia Support for 4:2:2 Color Space:
– Blackwell (RTX 50-Serie)

3. Intel Support for 4:2:2 Color Space
– 10th Gen and newer processors offer support for Hardware Accelerated encoding and decoding of HEVC codec on 4:2:2 color sampling via Quick Sync.
– Previous graphics controllers on 9th Generation Intel® Core™ Processors and older were limited to HEVC codec 4:2:0 color sampling for hardware accelerated encoding and decoding.

This is a quick config sample to encode 4:2:2 with Intel Quick Sync (10th Gen+)

```
format='dvb' protocol='udp://' video::codec='hevc_qsv' video::b='5M'
audio::codec='aac' audio::b='128K' video::pix_fmt='yuyv422'
```

Note: Features such as interlace and 4:2:2 color space are also available in x264 CPU encoders. Since we use an LGPL version of FFmpeg without GPL parts (e.g., libx264), these features are not enabled. It is possible to replace the FFmpeg libraries with your own versions.

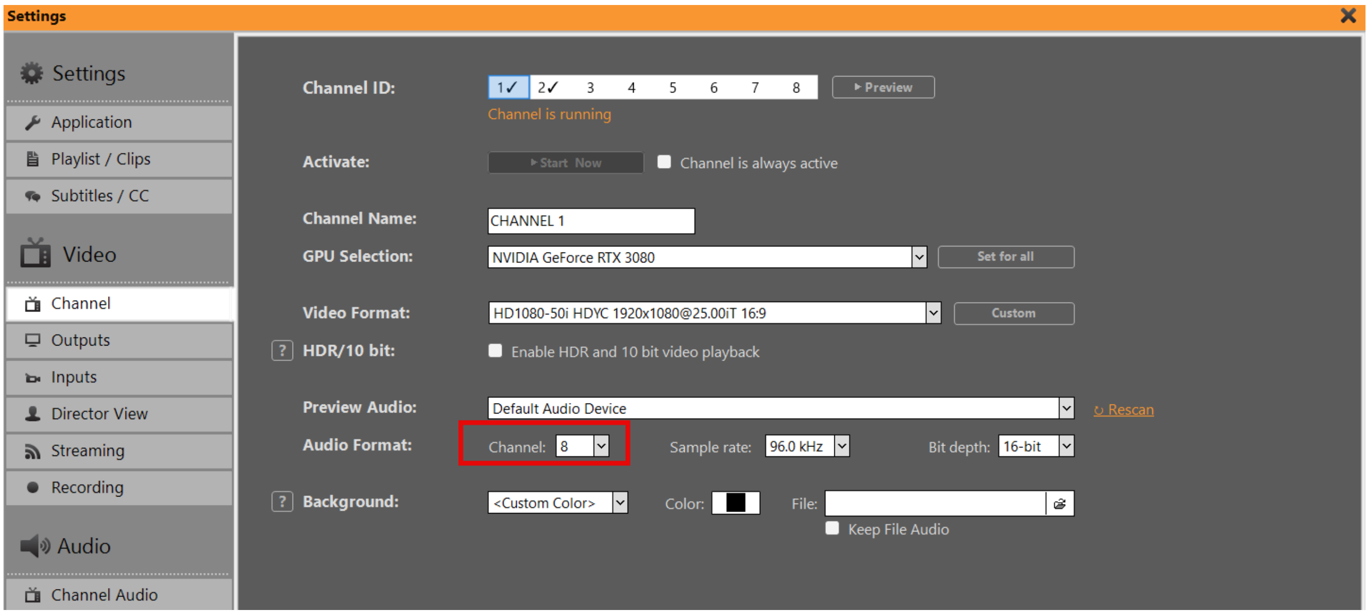
---

6. Multiple Audio Tracks

If you need to send multiple Audio Tracks, e.g. for different Languages or for Director Audio, you can do so with all supported Protocols (DVB, RTMP, UDP and some more). You can also apply different Audio Codecs.

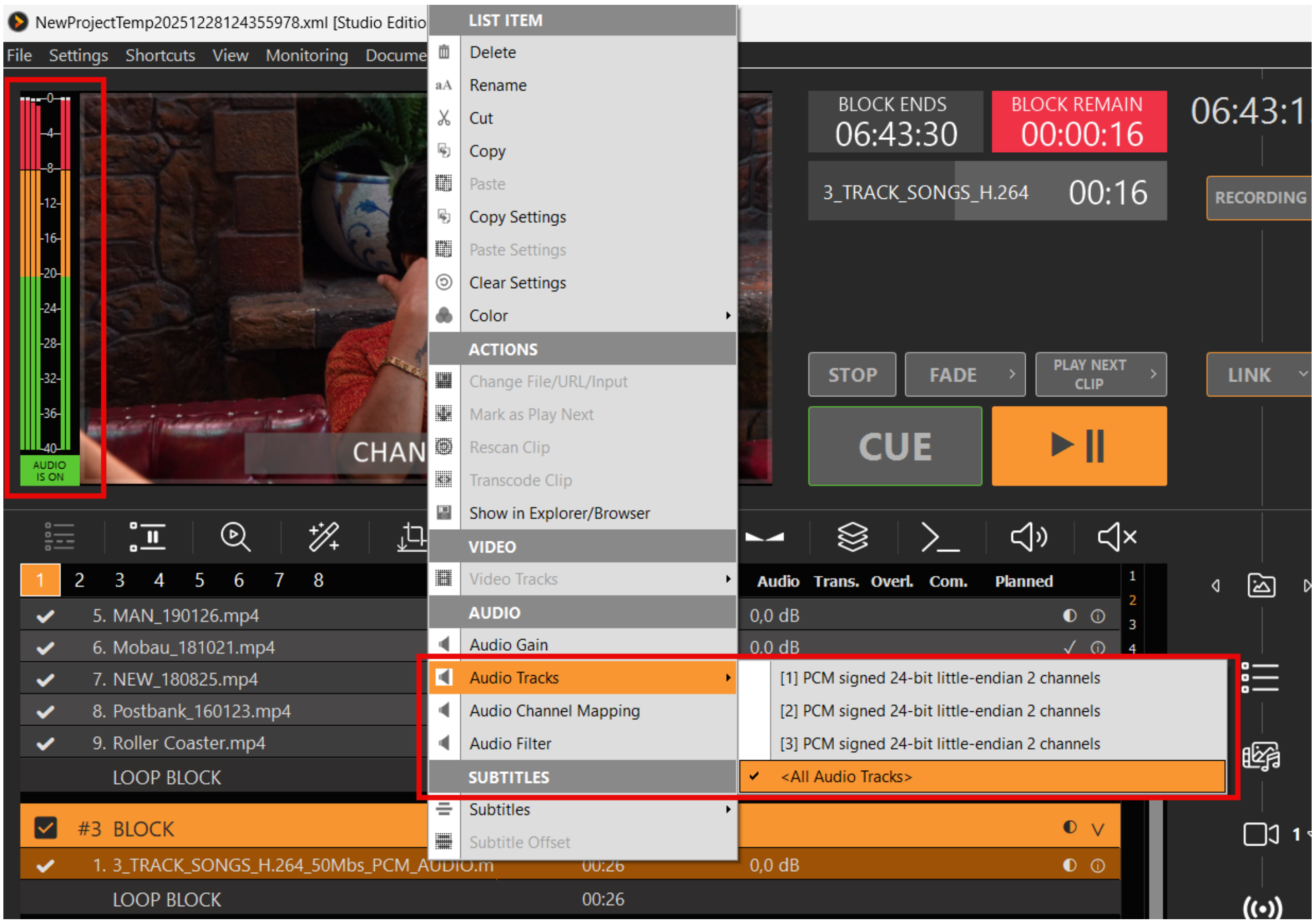
The way this works is, that you have ALL Audio Tracks already active in your Channel, then SPLIT the Audio Tracks for your Stream Output in e.g. Stereo Pairs.

First, make sure your Channel is set to more then 2 Audio Channel:

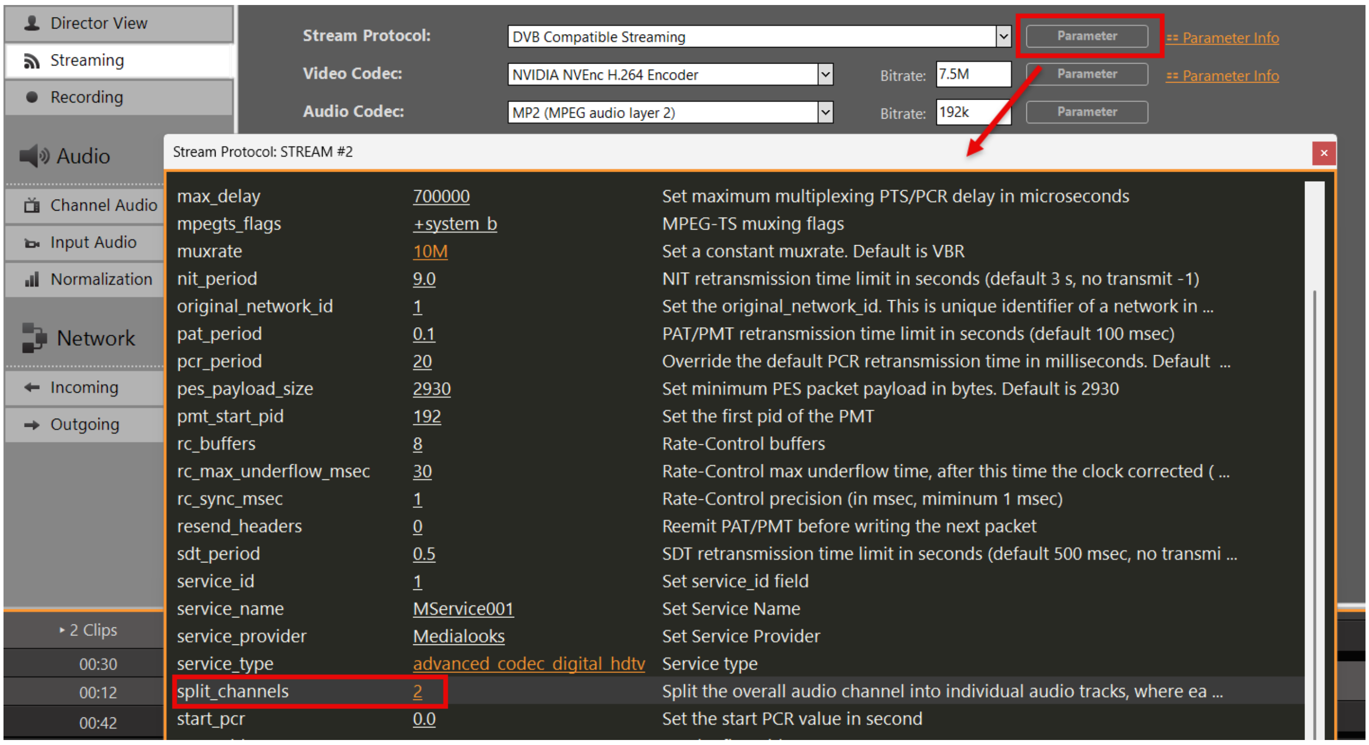


Channel ID: 1✓ 2✓ 3 4 5 6 7 8
Channel is running
Activate: Channel is always active
Channel Name: CHANNEL 1
GPU Selection: NVIDIA GeForce RTX 3080
Video Format: HD1080-50i HDYC 1920x1080@25.00iT 16:9
HDR/10 bit: Enable HDR and 10 bit video playback
Preview Audio: Default Audio Device
Audio Format: Channel: 8 Sample rate: 96.0 kHz Bit depth: 16-bit
Background: <Custom Color> Color: ■ File:
Keep File Audio

Then “fill” these Audio Channel with Audio Content, e.g. by enabling All Audio Tracks on a Clip or ingesting all Tracks via SDI (up to 16 Audio Tracks). You can also use Audio Mapping to mix up/down your Audio Tracks or use Audio Filter to copy and delay certain Audio Tracks.



Then you open your Stream Settings, especially the Parameter of your Stream Protocol and enable Audio Track Splitting:



Stream Protocol: DVB Compatible Streaming
Video Codec: NVIDIA NVEnc H.264 Encoder Bitrate: 7.5M
Audio Codec: MP2 (MPEG audio layer 2) Bitrate: 192k

Stream Protocol: STREAM #2

max_delay	700000	Set maximum multiplexing PTS/PCR delay in microseconds
mpegts_flags	+system_b	MPEG-TS muxing flags
muxrate	10M	Set a constant muxrate. Default is VBR
nit_period	9.0	NIT retransmission time limit in seconds (default 3 s, no transmit -1)
original_network_id	1	Set the original_network_id. This is unique identifier of a network in ...
pat_period	0.1	PAT/PMT retransmission time limit in seconds (default 100 msec)
pcr_period	20	Override the default PCR retransmission time in milliseconds. Default ...
pes_payload_size	2930	Set minimum PES packet payload in bytes. Default is 2930
pmt_start_pid	192	Set the first pid of the PMT
rc_buffers	8	Rate-Control buffers
rc_max_underflow_msec	30	Rate-Control max underflow time, after this time the clock corrected (...
rc_sync_msec	1	Rate-Control precision (in msec, miminum 1 msec)
resend_headers	0	Reemit PAT/PMT before writing the next packet
sdt_period	0.5	SDT retransmission time limit in seconds (default 500 msec, no transmi ...
service_id	1	Set service_id field
service_name	MService001	Set Service Name
service_provider	Medialooks	Set Service Provider
service_type	advanced_codec_digital_hdtv	Service type
split_channels	2	Split the overall audio channel into individual audio tracks, where ea ...
start_pcr	0.0	Set the start PCR value in second

INFO: The Number of Audio Channel in your Stream Settings need to stay at 2.

If needed, you can apply different Audio Codecs for the addition Audio

Tracks. Please note the added and increasing Numbers for every Audio Tracks after the first one:

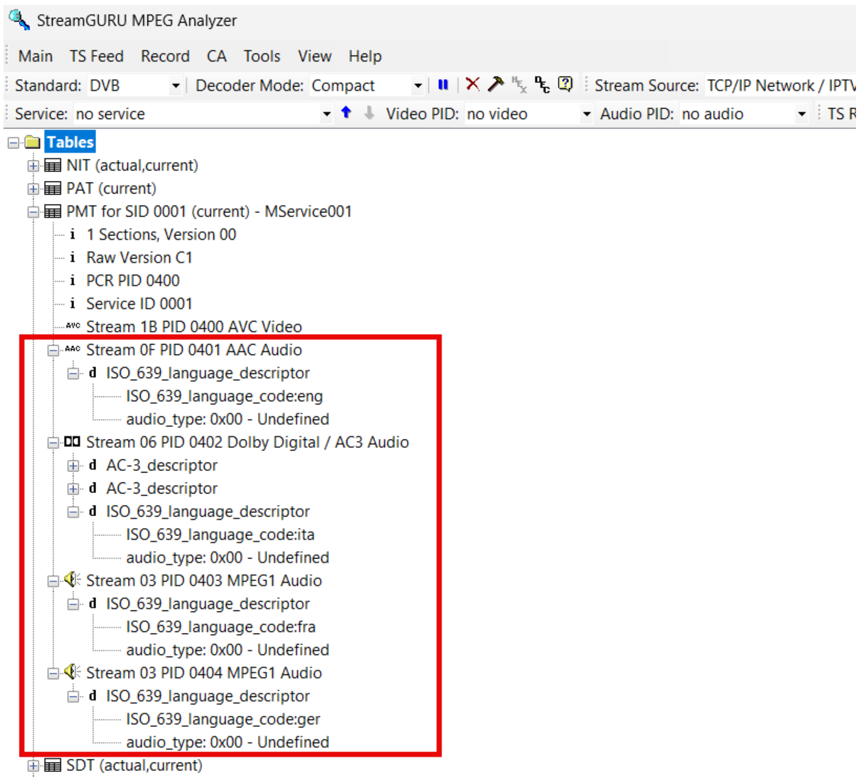
```
audio::codec='aac' audio.1::codec='ac3' audio.2::codec='libmp3lame'
audio.3::codec='mp2'
```

You can also apply a Language Descriptor for every Audio Track:

```
audio::metadata::language='eng' audio.1::metadata::language='ita'
audio.2::metadata::language='fra' audio.3::metadata::language='ger'
```

INFO: Language Codes must comply to ISO 639-2

You can then test your resulting Audio Tracks in e.g. StreamGURU:

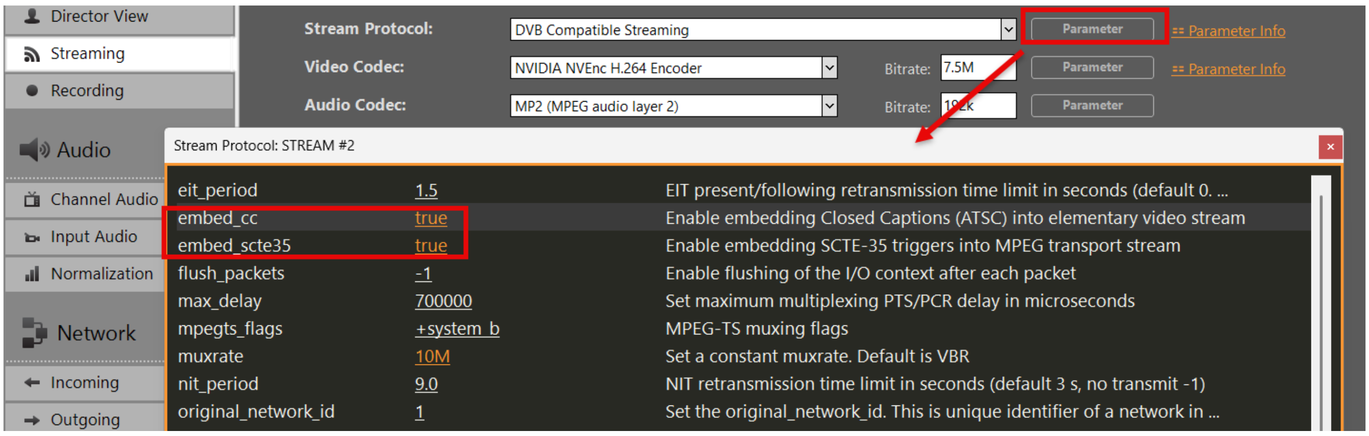

StreamGURU MPEG Analyzer
Main TS Feed Record CA Tools View Help
Standard: DVB Decoder Mode: Compact Stream Source: TCP/IP Network / IPTV
Service: no service Video PID: no video Audio PID: no audio

```
Tables
  NIT (actual,current)
  PAT (current)
  PMT for SID 0001 (current) - MService001
    1 Sections, Version 00
    Raw Version C1
    PCR PID 0400
    Service ID 0001
    Stream 1B PID 0400 AVC Video
    Stream 0F PID 0401 AAC Audio
      ISO_639_language_descriptor
        ISO_639_language_code:eng
        audio_type: 0x00 - Undefined
    Stream 06 PID 0402 Dolby Digital / AC3 Audio
      AC-3_descriptor
      AC-3_descriptor
      ISO_639_language_descriptor
        ISO_639_language_code:ita
        audio_type: 0x00 - Undefined
    Stream 03 PID 0403 MPEG1 Audio
      ISO_639_language_descriptor
        ISO_639_language_code:fra
        audio_type: 0x00 - Undefined
    Stream 03 PID 0404 MPEG1 Audio
      ISO_639_language_descriptor
        ISO_639_language_code:ger
        audio_type: 0x00 - Undefined
  SDT (actual,current)
```

---

7. Closed Captions and SCTE

Both can be enabled in the Parameter Settings of your Protocol:



Stream Protocol: DVB Compatible Streaming
Video Codec: NVIDIA NVEnc H.264 Encoder Bitrate: 7.5M
Audio Codec: MP2 (MPEG audio layer 2) Bitrate: 192k

Stream Protocol: STREAM #2

eit_period	1.5	EIT present/following retransmission time limit in seconds (default 0. ...
embed_cc	true	Enable embedding Closed Captions (ATSC) into elementary video stream
embed_scte35	true	Enable embedding SCTE-35 triggers into MPEG transport stream
flush_packets	-1	Enable flushing of the I/O context after each packet
max_delay	700000	Set maximum multiplexing PTS/PCR delay in microseconds
mpegts_flags	+system_b	MPEG-TS muxing flags
muxrate	10M	Set a constant muxrate. Default is VBR
nit_period	9.0	NIT retransmission time limit in seconds (default 3 s, no transmit -1)
original_network_id	1	Set the original_network_id. This is unique identifier of a network in ...

INFO: SCTE-35 is only available for Protocols: SRT, DVB, UDP (any Video Codec). Closed Caption will only work with Video Codecs: MPEG-2, H.264 (any Protocol)

This will require both to be enabled and available in your Channel already. Please refer to the following articles for more information on SCTE and/or Closed Captions:

→ Sending SCTE-35 to Stream Server
→ Closed Captions / Subtitles

---

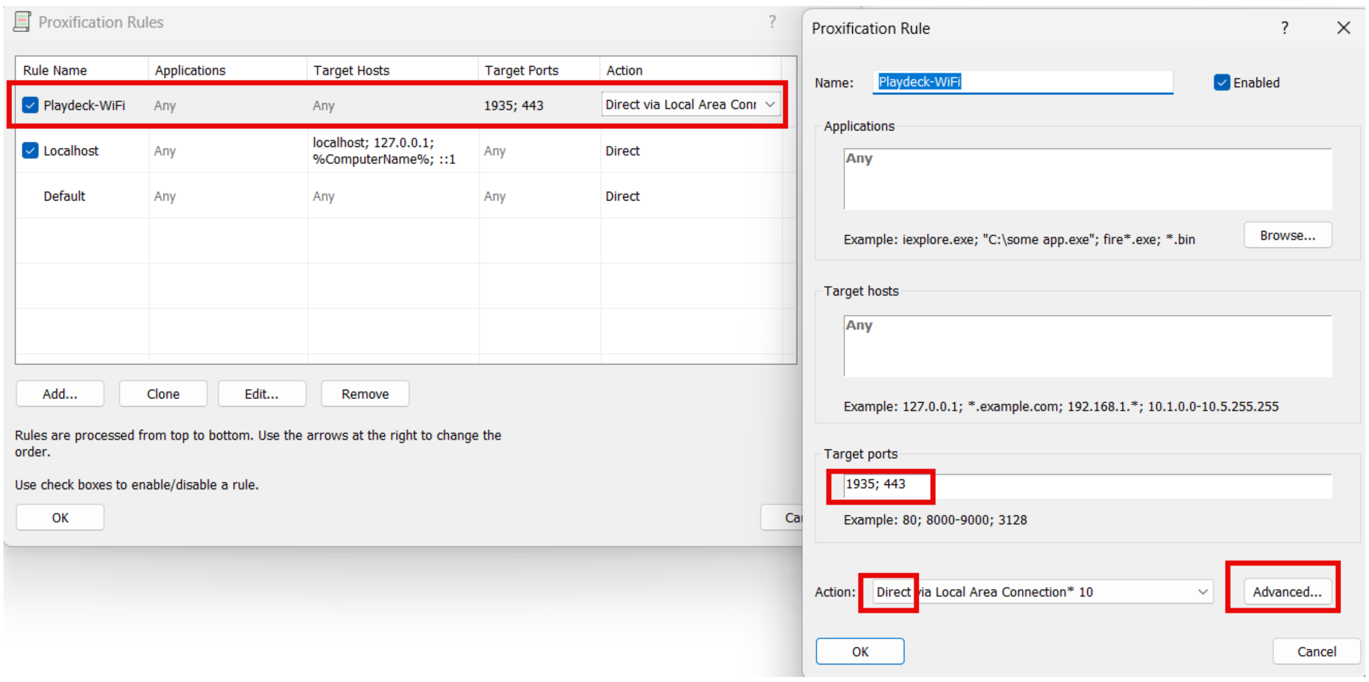
8. Routing Streams over specific IP

Your production requires you to send your streams via specific networks. This is mostly used to have a dedicated/reserved line for streaming, so it doesn't interfere with regular network traffic.

If using **UDP Streams** you simple add this parameter directly to the URL:
udp://ip:port?localaddr=172.20.10.2

If using **SRT Streams** you simple add this parameter directly to the URL. This only works in **rendezvous mode**:
srt://ip:port?mode=rendezvous&localip=172.20.10.2

If using RTMP Streams, there sadly is no paramter. You need to route all RTMP outbound traffic manually. For this we use Proxifier. Goto Profile, Proxification Rules, Add new Rule and edit like this and move it to the top if the rules. Pick your desired adapter via "Advanced".


Proxification Rules

Rule Name	Applications	Target Hosts	Target Ports	Action
✓ Playdeck-WiFi	Any	Any	1935; 443	Direct via Local Area Conn
✓ Localhost	Any	localhost; 127.0.0.1; %ComputerName%; ::1	Any	Direct
Default	Any	Any	Any	Direct

Rules are processed from top to bottom. Use the arrows at the right to change the order.

Use check boxes to enable/disable a rule.

Proxification Rule

Name: Playdeck-WiFi ✓ Enabled

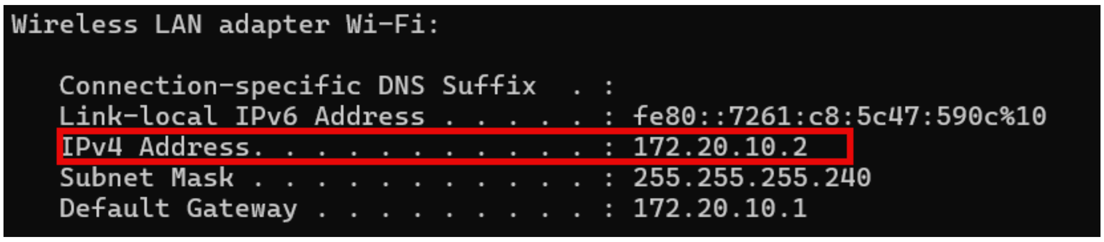
Applications: Any
Example: iexplore.exe; "C:\some app.exe"; fire*.exe; *.bin

Target hosts: Any
Example: 127.0.0.1; *.example.com; 192.168.1.*; 10.1.0.0-10.5.255.255

Target ports: 1935; 443
Example: 80; 8000-9000; 3128

Action: Direct via Local Area Connection* 10

If you want to run a dry test locally without the 2nd line, enable both Ethernet Adapter and Wifi on your PC, then Start a new Hotspot on your Mobile Phone and connect PC Wifi to that Phone. Then open Console ("cmd") and type “ipconfig”. The IP address you need is there:



```
Wireless LAN adapter Wi-Fi:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::7261:c8:5c47:590c%10
   IPv4 Address. . . . . . . . . . . : 172.20.10.2
   Subnet Mask . . . . . . . . . . . : 255.255.255.240
   Default Gateway . . . . . . . . . : 172.20.10.1
```

You can test your network traffic with wireshark.

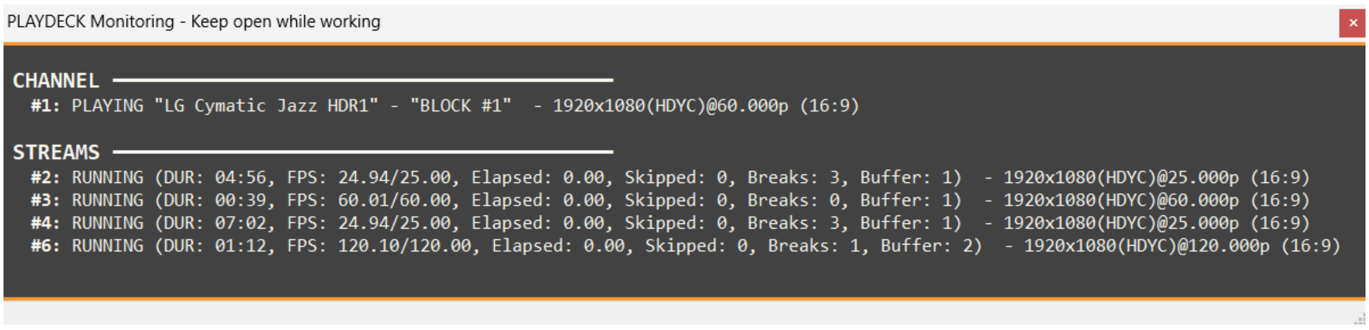
---

9. Monitoring & Troubleshooting

If your Streams are lagging or even stopping completely, you are often in the dark what exactly happened. Here are some pointer to help you identify the problem.

1. Monitoring

Open the Monitor Window in PLAYDECK or open the Stream Logs.



```
PLAYDECK Monitoring - Keep open while working

CHANNEL
  #1: PLAYING "LG Cymatic Jazz HDR1" - "BLOCK #1"  - 1920x1080(HDYC)@60.000p (16:9)

STREAMS
  #2: RUNNING (DUR: 04:56, FPS: 24.94/25.00, Elapsed: 0.00, Skipped: 0, Breaks: 3, Buffer: 1)  - 1920x1080(HDYC)@25.000p (16:9)
  #3: RUNNING (DUR: 00:39, FPS: 60.01/60.00, Elapsed: 0.00, Skipped: 0, Breaks: 0, Buffer: 1)  - 1920x1080(HDYC)@60.000p (16:9)
  #4: RUNNING (DUR: 07:02, FPS: 24.94/25.00, Elapsed: 0.00, Skipped: 0, Breaks: 3, Buffer: 1)  - 1920x1080(HDYC)@25.000p (16:9)
  #6: RUNNING (DUR: 01:12, FPS: 120.10/120.00, Elapsed: 0.00, Skipped: 0, Breaks: 1, Buffer: 2)  - 1920x1080(HDYC)@120.000p (16:9)
```

FPS 50/60 (FPS Average/Desired)
Compares the current and average encoding frame rate against the desired level. A drop here indicates the system cannot keep up with real-time encoding.

Elapsed (Frame Waiting Time in Seconds)
Shows Time elapsed after the last Frame has been processed. If this value increases, not frames can be processed due to missing Source, or false settings or no connection.

Skipped (Dropped Frames)
The most critical indicator of performance bottlenecks. If this value increases, the encoder is intentionally dropping frames because it cannot process them fast enough.

Breaks (Frames lost)
If the network is slow, you see Breaks because the Server *wants* a new frame but it cannot get there in time (for protocols that requires acknowledgement like RTMP or SRT). Breaks can also happen if the Source does not deliver a frame in time.

Buffer (Frames waiting)
Monitors the number of frames waiting in the internal queue. A constantly growing buffer typically precedes skipped frames or Breaks. **High Skipped, Low Breaks**: This usually means your **CPU/GPU encoder** is too slow. **High Breaks, Low/Zero Skipped**: This usually means the **network or source** is the bottleneck. The writer is ready to encode but the frames aren't arriving or being "pulled" at the correct time intervals due to the connection lag.

2. Network/Bandwidth Problems

Turn of all your Streams, open your Windows Console ("cmd" in Start Menu) and type

```
tracert <hostname_or_ip>
```

Use your Target URLs Hostname or IP Address. If you see any Values >100 ms, this will indicate a slow connection to your Stream Target.

Also keep in mind, that while today Download Speeds with Fibre, Cable or DSL or mostly high speed (>100 Mbps), the Upload Speed could be very much slower than that. Use a Tool like Speedtest to determine your actual Upload Speed.

Also use Bandwidth Profiler on the Stream Server to check, if your desired Bitrate is also received on the Server. Most Provider or Software will provide with Tools to measure the received Bitrate. Ideally it is shown as Graph, which can point out any Gaps during the Transmission. If you have no Tool at Hand, we recommend streaming to YouTube, as it has a build-in Stream Analyzer, which is very useful to diagnose any instable connections.

In addition, please check with your Provider or Service, if the provide any LOCAL Stream Ingest URL's for your Region or Country. As an example, Twitch provides these Twitch Ingest Endpoints.