MultiDyne®
Video & Fiber Optic Systems

CONFIGURATION AND INSTALLATION MANUAL

MD9200-DEC
&
MD9200-DEC-OG

OTT Streaming Media Decoders

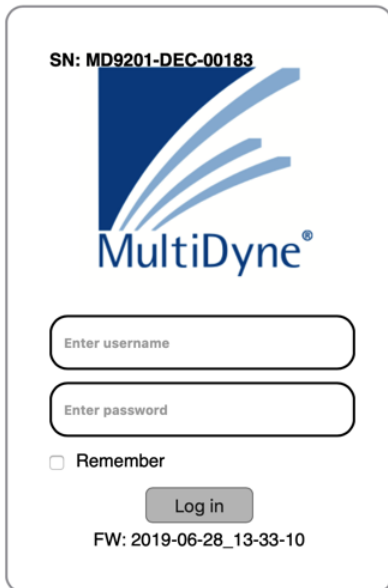**MD9200-DEC Software Manual**

Table of Contents

MD9200-DEC Software Manual

Logging into the User Interface

The MD9200-DEC uses a modern HTML5-embedded Web server for user configuration. Users can connect to the MULTIDYNE device via Gigabit Ethernet (GigE) port 1 or GigE port 2. GigE port 1 is factory defaulted to DHCP and GigE port 2 is factory defaulted to 192.168.2.7.

Gigabit Ethernet (GigE) port values

Gigabit Ethernet Port	Configuration
GigE 1	DHCP
GigE 2	192.168.2.7

Users are presented with the MULTIDYNE login screen when accessing the User interface, to which the **username** and **password** may be entered **(3)**. Currently usernames cannot be modified.

Username: 9200
Password: DEC

LEFT: The **unit serial number (1)** is displayed along the top of the login modal, and the current **firmware version** for the unit **(4)** is displayed along the bottom. To the right of the unit serial number, a **Help** link **(2)** may be accessed. This link will take users to the Help page, which provides more information about connecting to the UI, as well as unit technical information about the unit.

Once logged in, the MULTIDYNE unit UI is divided into two sections: the header, which is visible on all tabs, and the body, where the content depends on which tab is in use.

MD9200-DEC Header

ABOVE: The current tab in use is emboldened in **blue**.

Along the top left of the page are links to publicly accessible MULTIDYNE web pages, including the main website, Facebook, LinkedIn, MULTIDYNE Finder application in the Chrome store, and a **message envelope (1)**.

The **product type** is displayed in the top center **(2)**.

A **thumbnail (4)** is generated every five seconds to reflect the media being decoded by the unit.

The **Audio Group graph (5)** reflects levels of the channels as they update.

The **mini statistics graphs (6)** to the right display the rate of Mbps and, depending on the protocol, packets. Clicking the graph will resolve to the **STATISTICS** sub tab.

Along the top right of the page, **system time (3)** is displayed and updated per second, directly above a logout link.

The serial number and current **firmware version** of the unit **(7)** are displayed along the bottom right of the header, and the current **IPv4 address** of the unit on the left **(8)**.

When a transcode is running a blinking status bar will be visible for each one **(10)**.

The **status** of the decoder is displayed in the menu **(9)** and will change color as conditions change.

- When the player is up, or the UI is able to connect to and interact with the player, the box **(9)** will be **blue** and display DECODING. The **player status (11)** will be green.
- When the UI is experiencing difficulty interacting with the player the player status **(11)** will change to **yellow**, the status box **(9)** will do the same, displaying LISTENING. Both will resolve to blue or red after a short period of time.

- When the player is down, or cannot be connected to via the UI, the player status **(11)**

and the status box **(9)**, displaying OFFLINE, will turn **red**. When this occurs, a window (pictured below) appears informing users of the issue, shown below.
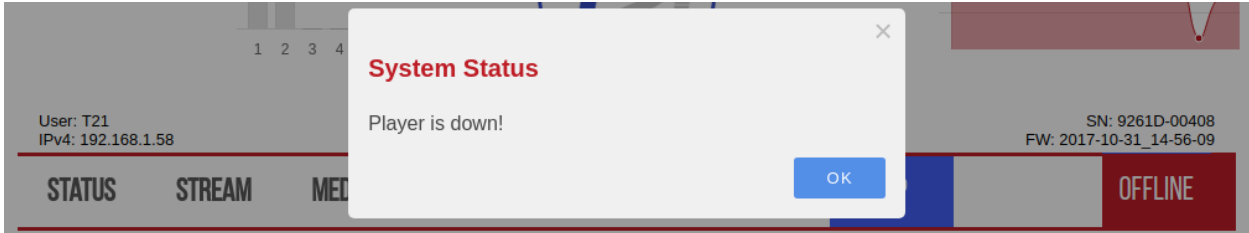
Users may click the **OK** button to exit the window, or it will disappear automatically when the issue is resolved.

Message Envelope

The envelope icon can be used to leave messages specific to how the unit is set up or what the unit is doing.
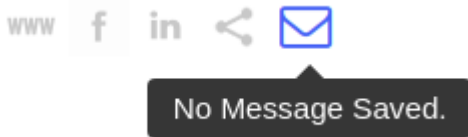
Hovering over the message icon when nothing has been saved will show this message or simply "message." Click to save a message.
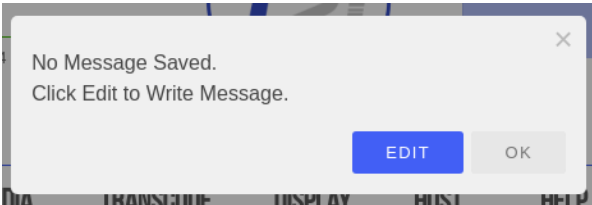
The first modal usually is a preview, or will give direction on how to add your message.

Enter your message and click save to apply your changes.

Now hovering over the message icon will preview your message.
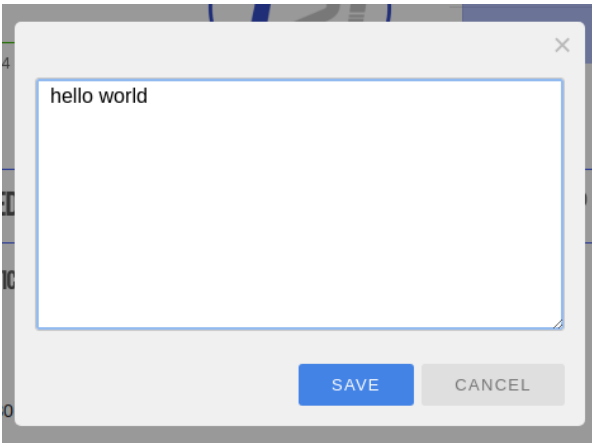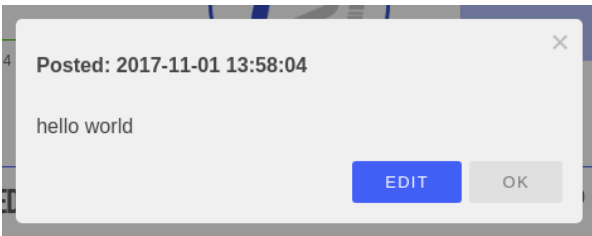

Clicking on the envelope will show the message and a timestamp of when it was posted.

STATUS

The STATUS Sub Tab

After successfully logging in to a unit, users are brought to the **STATUS** tab. Within the **STATUS** tab, information about the unit's current status is displayed, and program settings can be modified.

![MultiDyne - Video & Fiber Optic Systems]

MD9200-DEC Software Manual

STATUS	STREAM	MEDIA	TRANSCODE	DISPLAY	HOST	HELP	DECODING

STATUS	METADATA	STATISTICS

1 Restart Player

2 Input: udp://239.0.30.30:2000

3 Output: udp://239.0.58.58:2021

4 Transcode 1:   http://9261-00408.local/static/hls/stream1.m3u8

5 Transport: Playing Network - UDP

6 Source: Gigabit Ethernet 2

7 Current Play Time: 27 Minutes 52 Seconds

8 Program: Program 1   **9** Audio 1: 482 Format: LPCM Delay: 0

Video/Audio: Both Enabled   Edit Audio

Edit Video

ABOVE: The **Restart Player** button **(1)** will restart the current player without rebooting the device. Click **OK** on the resulting modal to begin the process.

The **Input (2)** displays the decoder's current input path.

The **Output (3)** displays the decoder's current output path.

The **Transcode (4)** displays the decoder's current transcode path. Each transcode that is outputting will display on this tab.

The **Transport (5)** refers to the type of media being decoded; this can be a network stream, media file on the unit, or playlist.

The **Source (6)** may be Gigabit Ethernet 1, Ethernet 2, or DISK.

The **Current Play Time (7)** is the amount of time the unit has been decoding the particular media.

The Video **(8)** information displays the **Program** and **Video/Audio**. These settings can be changed by clicking the **Edit Video** button.

The Audio **(9)** information displays the decoder's current **Audio pids**, **Format** and **Delay** which can be edited by clicking the **Edit Audio** button.
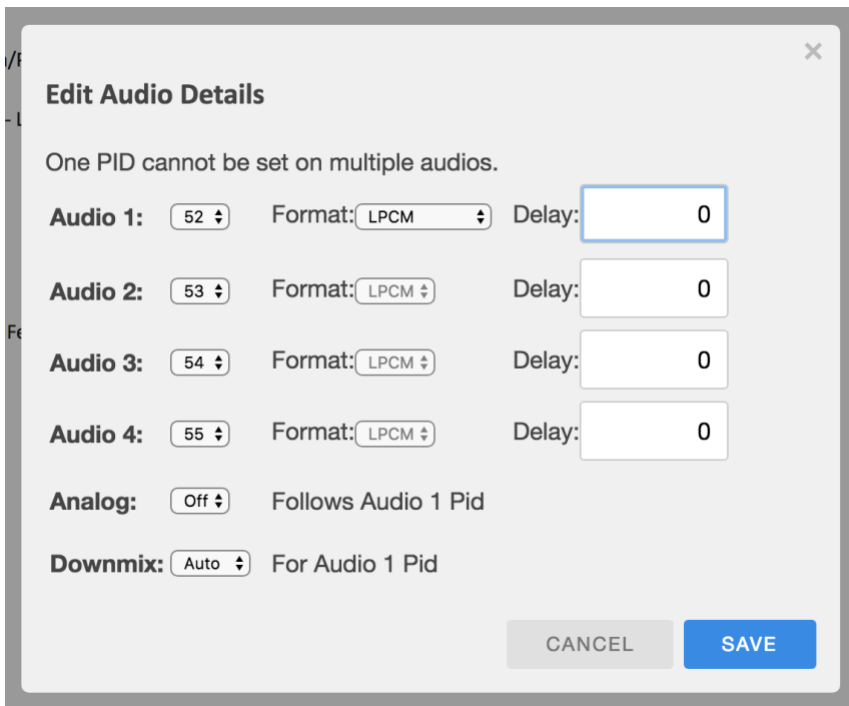
Edit Video
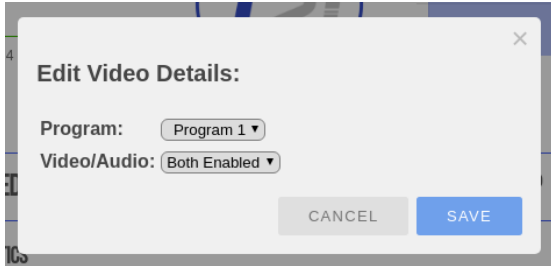
The **Program** determines which program number to play; audio options available vary depending on the media being decoded.
The **Video/Audio** determines whether the video, audio, or both is enabled. Clicking the **SAVE** button sets the changes.

Edit Audio

Audio 1-4 determine the current audio pid being enabled/decoded. Pids need to be set in consecutive order. **Format** determines the audio channel output format; options include LPCM (Linear Pulse Code Modulation), PASS SMPTE (Passthrough), and PASS ICE (Passthrough). **Delay** sets the audio offset, in milliseconds. Clicking the **SAVE** button sets the changes. Editing Audio Pid 1 or Audio 1 Format alone does not require the play path to be reacquired.

Program: Cband HD Home Feed	**Audio 1:** 52 **Format:** LPCM **Delay:** 0	
Video/Audio: Both Enabled	**Audio 2:** 53 **Format:** LPCM **Delay:** 0	
Edit Video	**Audio 3:** 54 **Format:** LPCM **Delay:** 0	
	Audio 4: 55 **Format:** LPCM **Delay:** 0	
	Edit Audio	
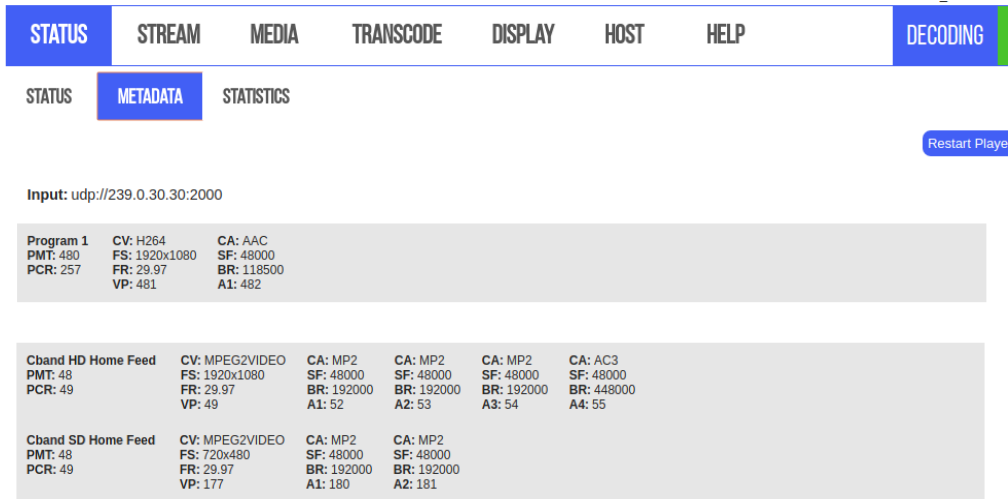
When more than one Audio Pid is available they will be displayed on the **STATUS** sub tab.

The METADATA Sub Tab

Shows any available metadata, by program, for an input stream being decoded. Can have information for as many programs as the stream contains.

The STATISTICS Sub Tab

ABOVE: The **STATISTICS** sub tab contains **information** about the input stream **(1)** updated in

intervals. The information available depends on what protocol is being decoded. All streams yield Mbps for the **Mbps graph (2)** but the **Packets graph (3)** will be available for certain protocols like UDP/RTP and Zixi streams.

STREAM

The INPUT Sub Tab

ABOVE: The **Ping Address (1)** may be used to determine whether or not a device, such as a source encoder, is on the same network as the MD9200-DEC. To use it, enter the IP address of the device in question. An Up or Down will be displayed next to this field indicating the status of the device.

The **History (2)** dropdown has the last ten streams that the unit successfully decoded. Click the **Clear History** button to erase them. This will halt decoding and restart the current player process without rebooting the device.

MD9200-DEC Software Manual

Choosing a different **protocol** will generate an appropriate form **(4)**. Make your changes and then click **Apply**.

Play Control buttons **(3)** start, stop, or play the next path if there are fallbacks applied. When Fallbacks are applied the **blue arrow (5)** becomes visible on the currently decoding stream. Clicking the **blue plus button** next to **Fallback (9)** will add up to three forms to apply a fallback. If a fallback source is configured, the MD9200-DEC unit will attempt to decode fallback sources in descending order until one is able to be successfully decoded. If the primary source stream becomes available after the player has switched to a fallback, the player will not be able to re-acquire the primary stream until the user re-applies the primary stream settings. Remove the fallback by clicking the **red X button** to the left.

Dejitter (6) determines the size of the dejitter buffer in milliseconds to allow for smoother decoding oversaturated networks and distances. A larger value corresponds to greater latency, so the number should not be higher than required for smooth decoding of the current source stream. The default setting of 33 milliseconds is sufficient for most networks.

Null Packet Detection (7) Software demux is normally used in null stripped sources. Setting to "disable", disables and forces hardware demuxer.

IGMP Filter Mode (8) settings can also be configured to allow for an inclusive or exclusive list of stream sources. If using a filter, the IGMP filter list will be shown. This can be used on networks with streams coming from many different sources, in order to prevent overlap of stream addresses. For example, if two encoders are both outputting a UDP stream at udp://239.0.1.125:2000, and it is known that the desired stream is coming from the encoder at address 192.168.2.7, an inclusive IGMP source list can be configured as demonstrated above for that specific encoder's IP address.

IGMP Filter Mode: Include ▾ [   ] Add

IGMP Address	Remove
192.168.2.7	⊖

Protocol   **URL**   **Stream ID** Regenerate
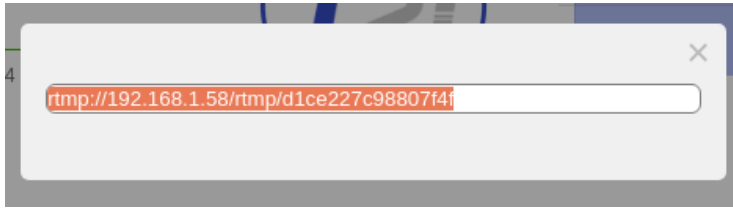
RTMP (Server) ▾   rtmp://192.168.1.58/rtmp/   d1ce227c98807f4f   Apply Copy Path

Clicking the **Regenerate** button will create a new RTMP **Stream ID** and replace the content in the field if there is any. Click the **Copy Path** button to open a modal with the full HLS path.

The OUTPUT Sub Tab

The Output Stream is a turnaround of the input.

Stream Fields

as of FW: 2017-10-31_14-56-09

FIELD NAME	DESCRIPTION	USE CASE
Protocol:	This determines the type of stream network protocol to transmit and/or receive. Each field is named for its given type, with some exceptions; the **Custom** protocol allows users to enter their own form of address, which will be parsed and utilized by the player. **Custom** is used with **HLS** and **RTSP**.	This field appears for every Protocol type.
Interface:	This field represents the physical ethernet port streams are received or transmitted on. The available options are Gig 1 and Gig 2.	**Input/Output/Transcode:** UDP, RTP, TCP
Address:	This field represents the IP address the stream will be decoding. No leading zeros in IPv4 addresses as we are not constrained to octals.	**Input/Output/Transcode:** UDP/RTP, SRT (Caller, Rendezvous), TCP, ZIXI (Pull, Feeder)
Port:	This field represents the ethernet port number for the stream.	**Input/Output/Transcode:** UDP/RTP, SRT (Caller, Listener, Rendezvous), TCP, ZIXI (Port, Accept, Feeder)
MTU:	**Maximum Transmission Unit** is the size of the largest network layer protocol data unit that can be communicated in a single network transaction.	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous)
Encryption / Password or Key:	Choose the encryption type, AES 128 (32 char) or AES 256 (64 char).	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous) **Output/Transcode:** ZIXI Feeder
Timeout:	Period in seconds that the connection will try before it hangs up. If left blank factory default is 10 seconds.	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous)
Latency:	This field represents the specified latency time for sending packets of data, given in	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous), ZIXI (Pull, Feeder)

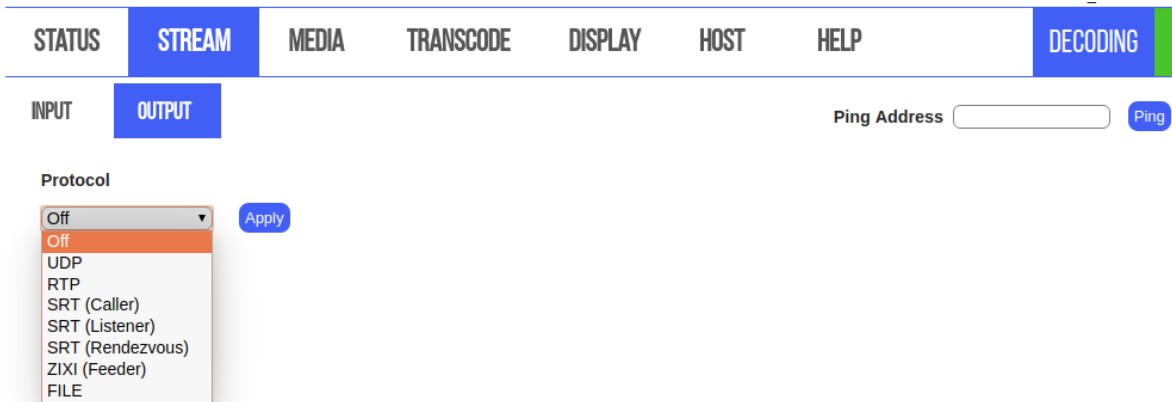|	milliseconds. The default latency value is 3000 milliseconds.	
Server Address:	IP address of the MULTIDYNE **FASP Server**.	**Input:** FASP Client
Target Rate:	Overall stream rate, matches the transport rate of the encoder profile.	**Input:** FASP Client
Remote ID / Password:	These fields represent the **Remote ID** and password of a Zixi broadcaster, which may be optional depending on the desired stream.	**Input:** ZIXI Pull
Stream ID:	This field represents the stream ID of a Zixi broadcaster and an **RTMP Server**. Clicking the **Regenerate** button will create a new RTMP **Stream ID** and replace the content in the field if there is any.	**Input:** ZIXI Pull, RTMP Server **Output/Transcode:** Zixi Feeder
URL:	This field represents the URL address of a stream. The **RTMP** fields is tied to its respective Protocol, while the **Custom** field allows users to enter an unspecified stream address that the decoder will parse. The **HLS** second field can be changed to a personalized **Stream ID**. Copy the whole path to receive the stream.	**Input:** RTMP (Client, Server), Custom **Transcode:** HLS
TTL:	**Time to Live** default number of hops between devices, default value is 64.	**Output/Transcode:** UDP/RTP, SRT (Caller, Rendezvous)
Traffic Shaping:	Buffers the output data to ensure constant intervals between packets.	**Output/Transcode:** UDP/RTP
Bandwidth Overhead:	Percentage used for forward error correction.	**Output/Transcode:** SRT (Caller, Rendezvous)
ToS:	**Type of Service** is the payload type for prioritizing traffic through routers.	**Output/Transcode:** SRT (Caller, Rendezvous)
Max Bitrate:	The rate of the transport output should be set equal or greater than the total output of the device. Recommended to set 20% higher.	**Output/Transcode:** ZIXI Feeder

Min Bitrate:	Setting a value enables Adaptive Bitrate.	Output/Transcode: ZIXI Feeder
Output Location:	List of available directories.	Output/Transcode: File

See the **MULTIDYNE Decoder Stream Configuration Guide** for more information.

MEDIA

The MEDIA Sub Tab
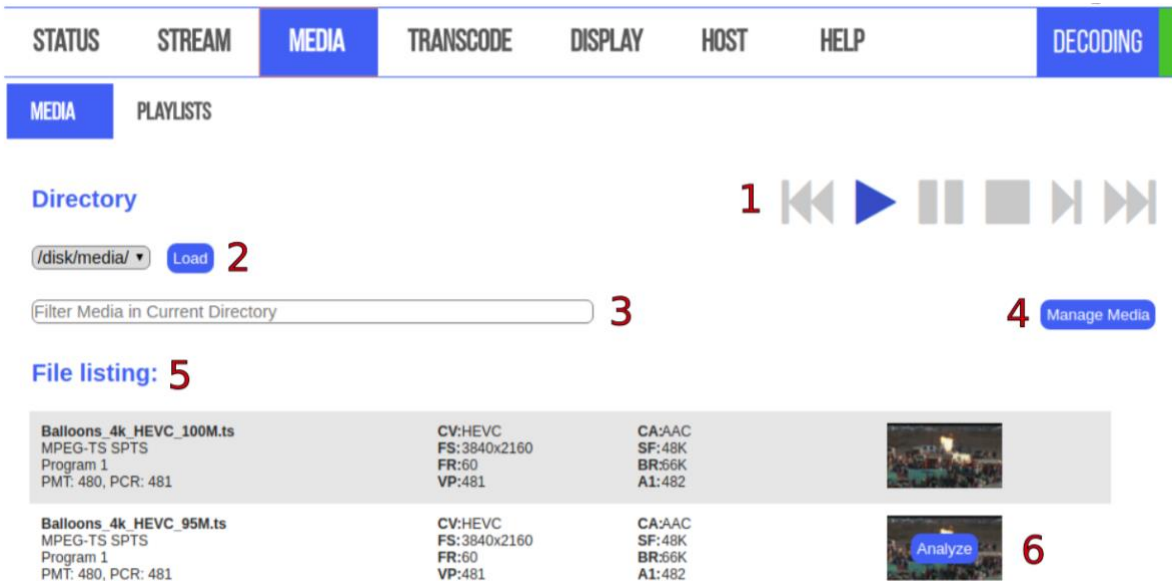
ABOVE: **Play Control** buttons **(1)** rewind, start, pause, stop, play the next path, or fast forward. If a video from media is being decoded, will cycle through media folder.

Load button **(2)** will start decoding a video and then cycle through the folder on loop. The **Filter (3)** searches the titles of the media loaded on the unit. Clicking the **Manage Media** button **(4)** will reveal tools to upload and edit media. When hovering over the thumbnail the **Analyze** button **(6)** shows. Clicking it brings up a modal with metadata for that file, shown to the LEFT.

MD9200-DEC Software Manual

Manage Media

ABOVE: Choose a file from your computer and click **Upload** **(1)** to copy it to the unit. Clicking the **Pencil** icon **(2)** brings up the Rename File modal (bottom left) and the **red X** icon **(3)** brings up the delete file modal (bottom right).

The PLAYLISTS Sub Tab

BELOW: See the **Media Play Control** for details **(1)**. Playlists are saved in the **Playlists** box **(2)**, click them to highlight and show contents in the **Playlist Media** box **(3)**. Double click a playlist title to begin playing.
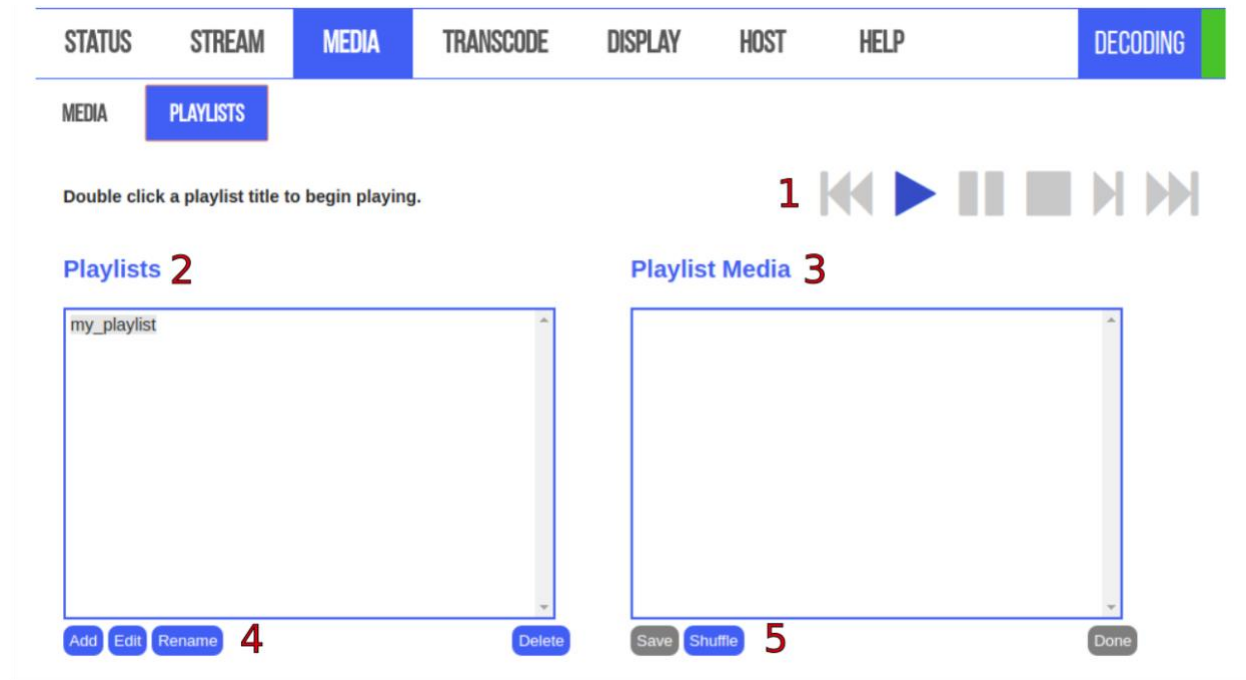
While the playlist is highlighted click the buttons **(4)** under the Playlists box to **Edit**, **Rename**, or **Delete** the playlist. **Add** a new playlist at any time.

MD9200-DEC Software Manual

Playlists

my_playlist

Add Edit Rename   Delete

Playlist Media

1 Big_Cats_h265_3840x2160_25000_5

Save Shuffle   Done

Media   Filter Media

1 Boats_h265_3840x2160_25000_59_1
2 Big_Cats_h265_3840x2160_25000_5
3 Coffee_h265_3840x2160_25000_59_
4 Balloons_4k_HEVC_95M.ts
5 Bridges_h265_3840x2160_25000_59
6 Balloons_4k_HEVC_100M.ts
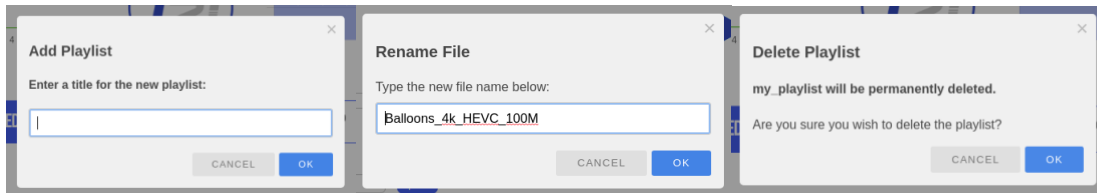
Add media to the playlist when you click **Edit** and the **Media** box will appears. Drag the media to the **Playlist Media** box and click **Save**. When there are no changes to apply the **Save** button will be disabled, as seen below. Click **Done** to hide the **Media** box again.

Playlist Media

1 Big_Cats_h265_3840x2160_25000_5
3 Coffee_h265_3840x2160_25000_59_

Save Shuffle   Done

Playlist Media

1 Big_Cats_h265_3840x2160_25000_5
3 Coffee_h265_3840x2160_25000_59_

Save Shuffle   Done
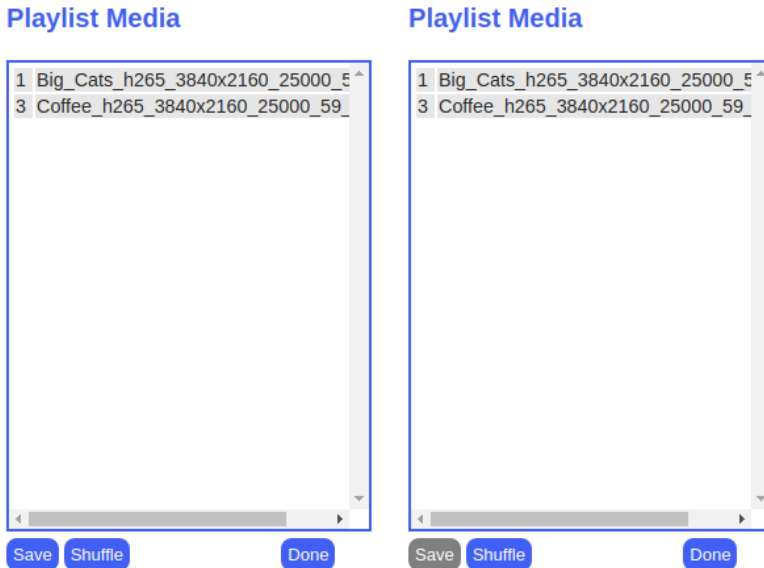
TRANSCODE

Available on decoders when feature is unlocked. Please contact support@MultiDyne.biz for information on purchasing feature keys.

The OUTPUTS Sub Tab
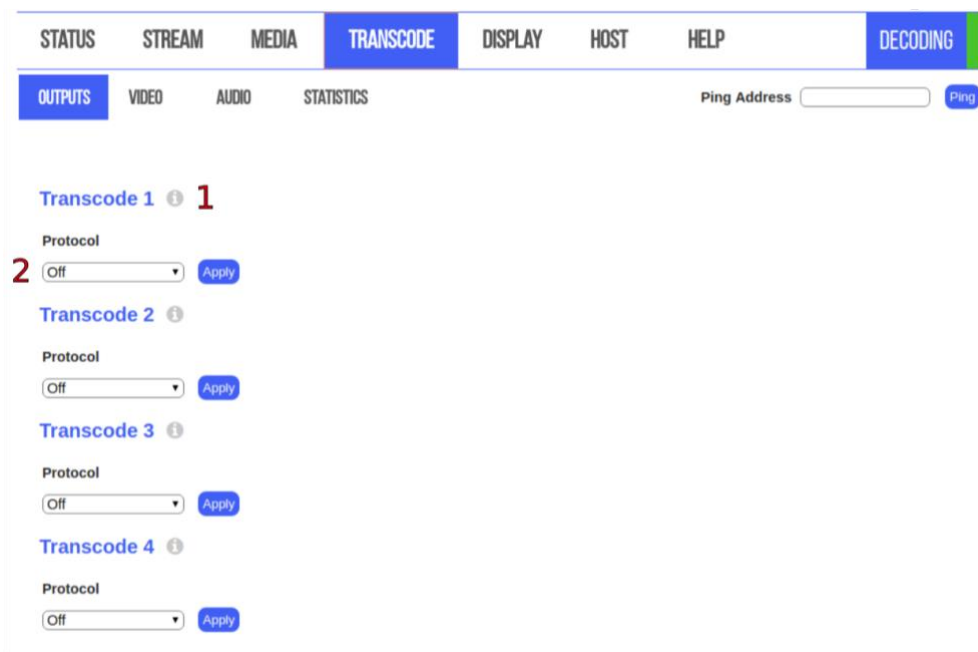
ABOVE: Transcoder allows user to make changes to output video and audio. Number of transcode streams available depend on the codec of the incoming stream. For example, HEVC allows only 2 transcodes to be set. When a transcoder is on a bar will be flashing in the status menu (RIGHT).

Output Status: Down

Video Settings:

PMT	Remux Mode	Remux Bitrate
480	constant	6000000
Video PID	**Video Mode**	**Video Bitrate**
481	constant	4000000
PCR PID	**PCR Interval**	**Ancillary**
257	30	Off
GOP	**Framing**	**GOP Size**

ABOVE: The **"i" icon (1)** next to each transcoder will bring up a modal (seen LEFT) with values currently configured. Values can be viewed regardless of whether an active output is configured.

BELOW: Changing the **Protocol (1)** will select a suggested **Profile (3)**. Clicking on the **Profile Title (2)** will resolve to the **Profile Form** with the corresponding profile selected. The **Audio Pid (4)** will be available when an audio profile is enabled. A single transcoder cannot have the same value assigned to both Audio Pids.
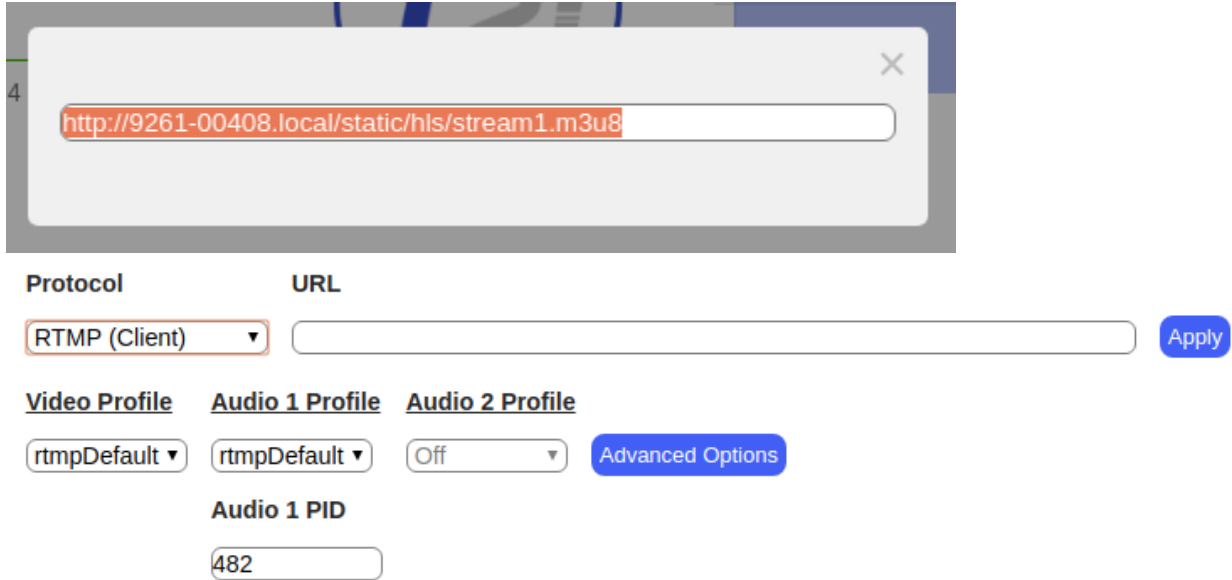
Click the **checkbox** next to **Min Bitrate** to enable the input and enable Adaptive Bitrate (ABR).

Click the **Copy Path** button to open a modal with the full HLS path.

MD9200-DEC Software Manual
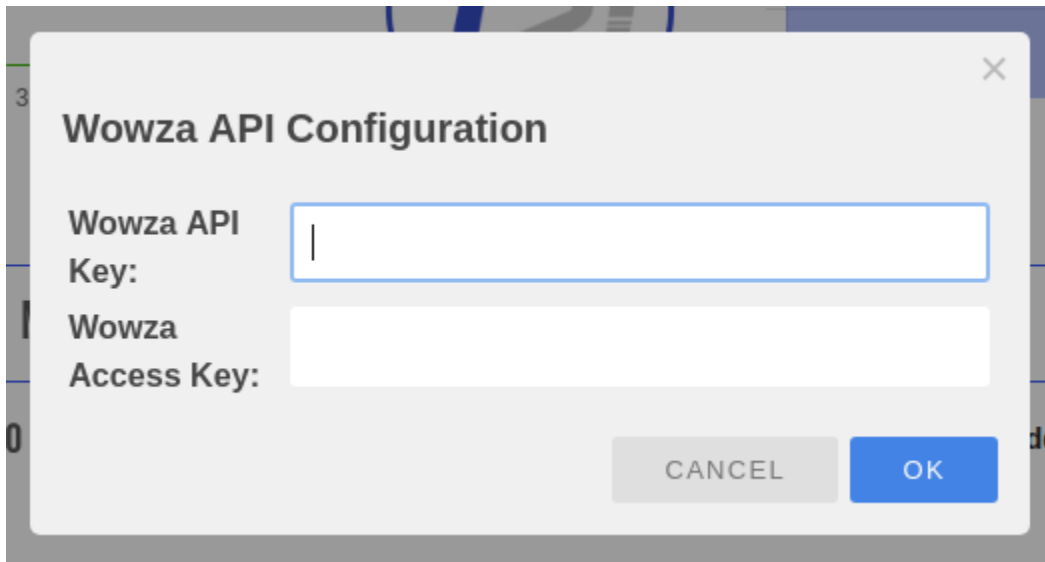
Click Advanced Options to bring up the Wowza modal. Contact support@MultiDyne.biz for information on purchasing keys.

The VIDEO Sub Tab

ABOVE: Default profiles will automatically populate in the **menu box (1)**. A selected profile will be highlighted **blue** and populate the **profile form (2)**. Any changes to a default profile need to be saved by clicking the **Update Profile** button **(3)**. To restore the default values to a default profile click the **Restore Defaults** button **(4)**. Default profiles cannot be renamed or deleted. Clicking the **New Profile** button **(5)** will allow the user to set a custom name and values.

LEFT: Click **Create Profile** button **(2)** to save custom profile. Leaving a field empty will prevent the profile from being saved. Spaces in the **Profile Name** will be parsed to underscores (_).

LEFT: When the custom profile is highlighted, the **Profile Name** field **(1)** is editable. **Update Profile (2)** saves changes. **Reset Changes (3)** restores values from the last save. **Delete (4)** deletes the profile. **New Profile (5)** generates the **Create Profile** state.

The AUDIO Sub Tab

ABOVE: Default profiles will automatically populate in the **menu box (1)**. A selected profile will be highlighted **blue** and populate the **profile form (2)**. Any changes to a default profile need to be saved by clicking the **Update Profile** button. To restore the default values to a default profile click the **Restore Defaults** button. Default profiles cannot be renamed or deleted. Clicking the **New Profile** button will allow the user to set a custom name and values.

Audio Profiles

default
hlsDefault
rtmpDefault

1 **Profile Name**

2 **Audio Pass**

Bitrate **Codec** **Profile**
128 kbps MPEG-4 AAC in ADTS AAC-LC

3 Create Profile

ABOVE: Click **Create Profile** button **(3)** to save custom profile. Leaving a field empty will prevent the profile from being saved. Spaces in the **Profile Name (1)** will be parsed to underscores (_). When **Audio Pass (2)** is not enabled more details are available.

Audio Profiles

default
hlsDefault
rtmpDefault
test

1 **Profile Name**

test

Audio Pass

Bitrate **Codec** **Profile**
128 kbps MPEG-4 AAC in ADTS AAC-LC

Update Profile Reset Changes Delete New Profile
2 **3** **4** **5**

ABOVE: When the custom profile is highlighted, the **Profile Name** field **(1)** is editable. **Update Profile (2)** saves changes. **Reset Changes (3)** restores values from the last save. **Delete (4)** deletes the profile. **New Profile (5)** generates the **Create Profile** state.

The STATISTICS Sub Tab

Transcode 1 Stats

Path: http://9261-00408.local/static/hls/stream1.m3u8
Elapsed Time: 4 Minutes 56 Seconds
Mbps: 2.37 Mbps
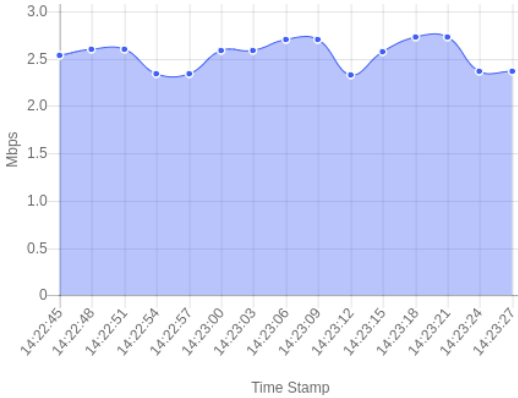
ABOVE: The Statistics Tab has graphs and details for each enabled transcoder updated on an interval.

DISPLAY

The FORMAT Sub Tab
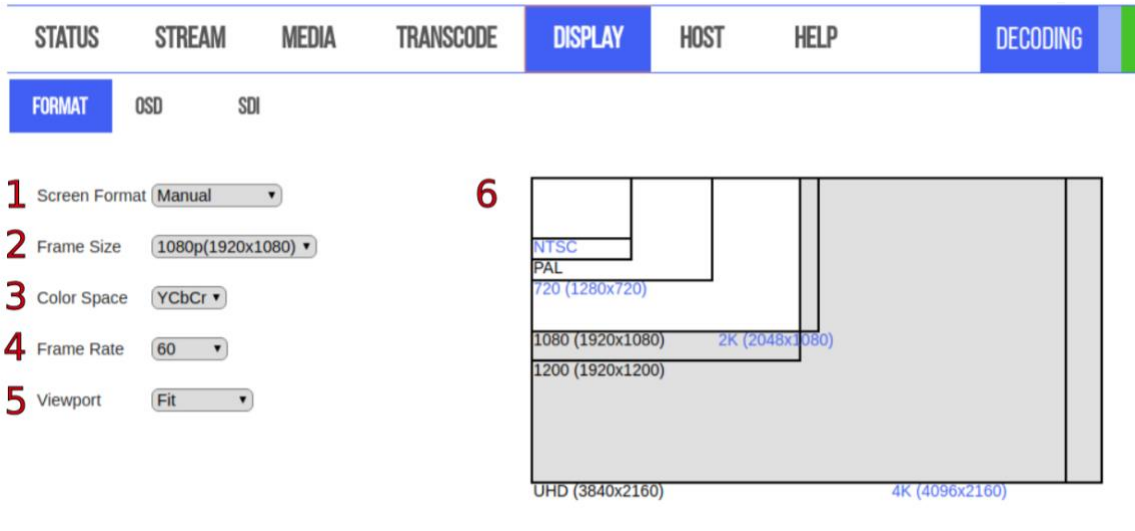
ABOVE: **Screen Format (1)** Manual changes the size of the video output to match your configured display. Follow mode follows the source format. **Frame Size (2)** for the decoder may be set to 4K, UHD, 1080p or 1080i, 720p, PALi or PALp, or NTSCi or NTSCp. Frame Size is represented in the graphic to the right **(6)**. **Color Space (3)** determines the color representation the decoder will use; options are YCbCr or RGB. **Frame Rate (4)** may be restricted depending on the frame size, and the output frame rate may depend on the capability of the media being decoded, regardless of frame rate selected. Frame rate options are 60, 59.94, 50, 30, 29.97, 25, 24, and 23.976 fps. Clicking **Viewport (5)** determines how the video is scaled to fit the display port. Current options are Original, Fit, Letter Box, Pan Scan, and Scale. When viewport is set to scale, users may also Scale By Percent.

The OSD Sub Tab

ABOVE: **Clicking Meta OSD (1)** allows users to toggle an overlay of information displayed on the screen. **Caption Service (2)** allows users to set visibility of a specific caption service. **Caption Display (3)** toggles the visibility of closed captions; current options include SDI closed captions, OSD closed captions, or both. Users may select a logo to display on the screen with **Logo Image (4)**. Any logo currently housed on the unit, in the /disk/logo directory, may be used. Users may add logos from their computer to the disk/directory by clicking **Upload Logo (5)**. Clicking **Signal Loss Behavior (6)** allows for the user to set player behavior upon stream decoding obstruction; options include Close (where the display is cleared and a black screen resolved), Hold frame, and Color with On-Screen Display (where a monochromatic screen is resolved, with overlaid stream information). If set to show an On-Screen Display, **Signal Loss Message (7)** allows for the user to determine the displayed message. **Signal Loss Frame Color (8)** sets the color for the raster that appears when the stream is halted.

The SDI Sub Tab

BELOW: The SDI subtab collects and displays status information about the decoding video. This tab and its information are only available for videos currently decoding in an SD mode.

HOST

The SYSTEM Sub Tab

ABOVE: When a user elects to **Reboot System (1)** cycles the unit off and back on. **Factory Reset (2)** restores the player configuration to default settings and restarts the player. Beneath the 'System Stats' header, system information may be obtained. **System Location (3)** sets the current physical location of the unit for later aid in locating. **Serial Number (4)**, **Firmware Version (5)**, and **SDK Version (6)** of the unit may also be viewed. **System Uptime (7)** displays the length of time the unit has been powered on since the last reboot or firmware update. Beneath the **Update Firmware** header **(8)**, users can update the current firmware version in use by the unit. To begin choose a file for upload. After uploading the firmware, the user has ten seconds to cancel before the update automatically begins. The **Samba Server (9)** idk. The **HTTP** dropdown **(10)** changes the domain to a secure server. **Disk Information (11)** displays to total amount of disk memory, the amount used, and the amount of free memory. This information is displayed both via image and in text for the user. Within the **Timezone** heading **(12)**, the user may update the unit's current time zone location and the NTP server address. Here, the time

zones are divided into larger categories by region. Choosing a regional category will provide the user with options specific to their category to choose from.

Update Password **1**

Old Password

New Password

Copy of New Password

Update Password

Create User **2**

Username

Password

Copy of Password
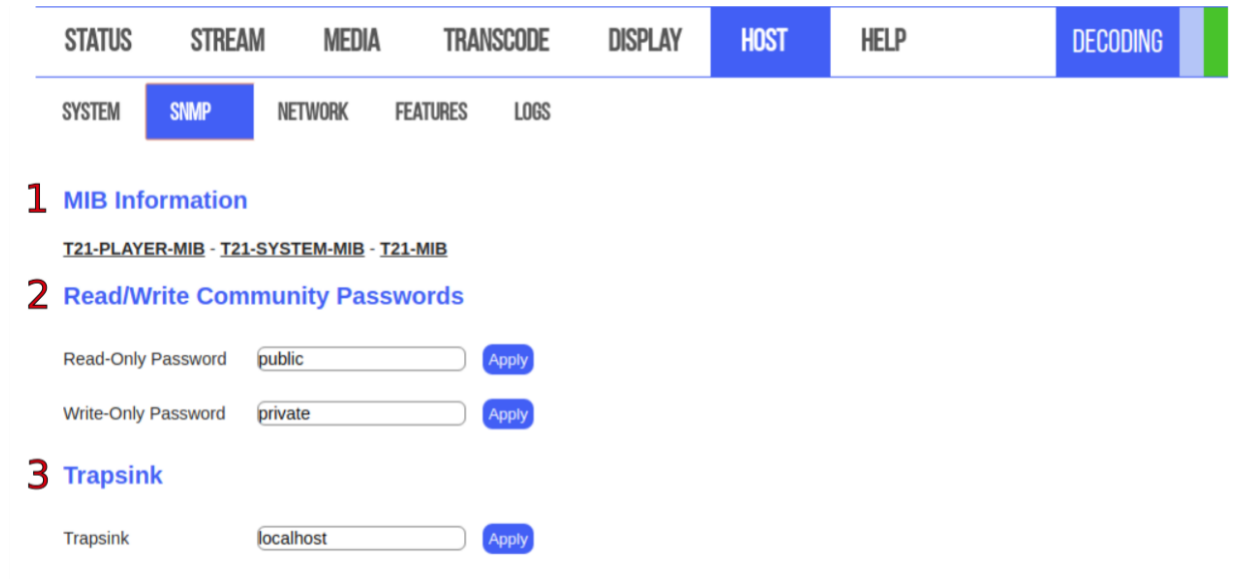
Create User

LEFT: (Bottom of the **SYSTEM** tab) **(1)** Update the password of the user currently signed in as. **(2)** Create a new user for that unit.

The SNMP Sub Tab

From the SNMP subtab, users may view and update community passwords, as well as the trapsink, and view current fields of the player MIB, system MIB, and main MIB.

BELOW: **(1) MIB Information** Links will take users to a text version of the MIB chosen. **(2) Read/Write Community Passwords** correlate to the privacy settings for SNMP GET/SET requests. **(3) Trapsink** specifies the host to send SNMP traps to.

The NETWORK Sub Tab

ABOVE: **(1) Zero Conf Interface** specifies the GigE interface the localhost will resolve to. Network interface configurations may be modified in the **Gigabit Ethernet (2/3)** sections. **Connection status** is marked by **Up** or **Down**, depending on availability. **DHCP** may also be toggled. **IPv4 Address**, **IPv4 Subnet Mask**, **Gateway**, **IPv6 Address**, **MAC Address** may also be set in this subtab. **Link Speed** is also displayed. **(4) Wireless** can be configured to DHCP or Static if one has been installed. **(5) Factory Restore** defaults the network settings and restarts the network.

BELOW: **DNS** settings **(1)** of the unit may be viewed. **Route** settings **(2)** for the network may also be set; available interface options are Gig 1 and Gig 2. For most uses, however, the default route is used. 'Zero Conf' interface may be set to GigE1 or GigE2.

1 DNS

DNS 1   DNS 2

192.168.1.1
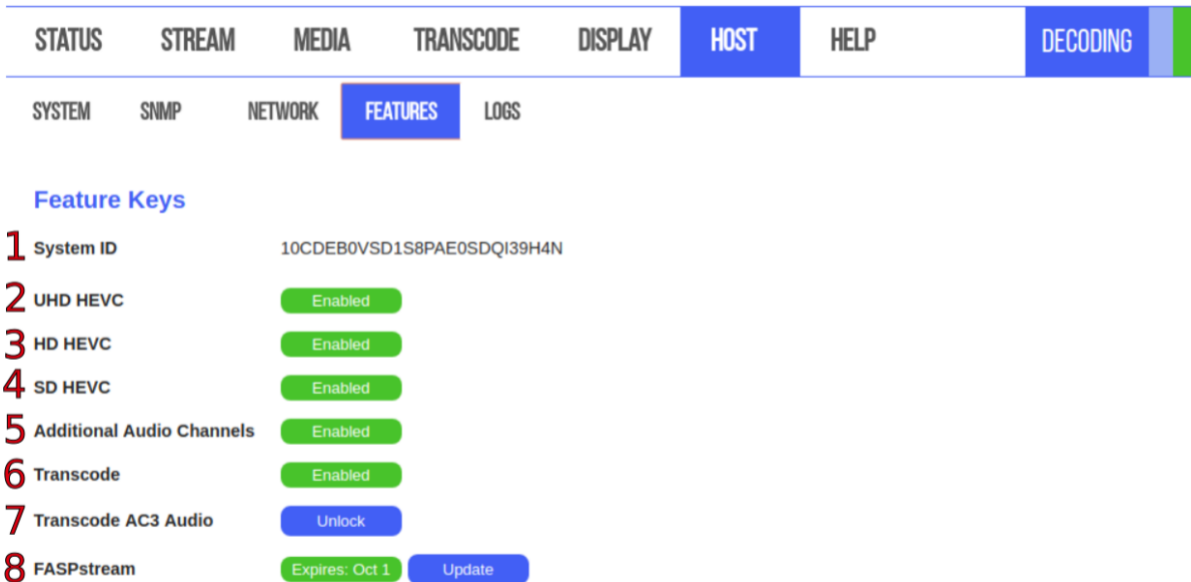
Apply

2 Route

Interface   Address   Mask   Gateway

Apply

The FEATURES Sub Tab

| STATUS | STREAM | MEDIA | TRANSCODE | DISPLAY | HOST | HELP | DECODING |

| SYSTEM | SNMP | NETWORK | FEATURES | LOGS |

Feature Keys

1 System ID   10CDEB0VSD1S8PAE0SDQI39H4N

2 UHD HEVC   Enabled

3 HD HEVC   Enabled

4 SD HEVC   Enabled

5 Additional Audio Channels   Enabled

6 Transcode   Enabled

7 Transcode AC3 Audio   Unlock

8 FASPstream   Expires: Oct 1   Update

ABOVE: To unlock a Feature, users need to provide the **System ID (1)**. **UHD HEVC (2), HD HEVC (3), SD HEVC (4)** allow users to decode those video codecs. **Additional Audio Channels (5)** allows up to four audios. **Transcode (6)** unlocks the feature allowing users to edit the video and audio of multiple outputs. **Transcode AC3 Audio (7)** is only visible if Transcode has been unlocked. Allows users to transcode audio into AC3 format. **FASPstream (8)** unlocks the FASP protocol from ASPERA.

The LOGS Sub Tab

BELOW: **Download Log button (1)** will download a tar file to user's computer. **Player** button **(2)** will show logs from the player. **System** button **(3)** will show all logs from the system. **All** button **(4)** will show all logs. The selected **Filter** button will be white with a **blue** border. Scroll to the bottom of the logs to see most recent.

ENCODER STREAM CONFIGURATION GUIDE

MD9200-DEC
&
MD9200-DEC-OG

OTT Streaming Media Decoders

MD9200-DEC Software Manual

Table of Contents

Stream Fields

FIELD NAME	DESCRIPTION	USE CASE
Protocol:	This determines the type of stream network protocol to transmit and/or receive. Each field is named for its given type, with some exceptions; the **Custom** protocol allows users to enter their own form of address, which will be parsed and utilized by the player. **Custom** is used with **HLS** and **RTSP**.	This field appears for every Protocol type.
Interface:	This field represents the physical ethernet port streams are received or transmitted on. The available options are Gig 1 and Gig 2.	**Input/Output/Transcode:** UDP, RTP, TCP
Address:	This field represents the IP address the stream will be decoding. No leading zeros in IPv4 addresses as we are not constrained to octals.	**Input/Output/Transcode:** UDP/RTP, SRT (Caller, Rendezvous), TCP, ZIXI (Pull, Feeder)
Port:	This field represents the ethernet port number for the stream.	**Input/Output/Transcode:** UDP/RTP, SRT (Caller, Listener, Rendezvous), TCP, ZIXI (Port, Accept, Feeder)
MTU:	**Maximum Transmission Unit** is the size of the largest network layer protocol data unit that can be communicated in a single network transaction.	**Output/Transcode:** SRT (Caller, Listener, Rendezvous)
Encryption / Password or Key:	Choose the encryption type, AES 128 (32 char) or AES 256 (64 char), which will be used on an entered password.	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous), ZIXI (Pull, Accept) **Output/Transcode:** ZIXI Feeder
Timeout:	Period in seconds that the connection will try before it hangs up. If left blank factory default is 10 seconds.	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous)
Latency:	This field represents the specified latency time for sending packets of data, given in milliseconds. The default latency value is 3000 milliseconds.	**Input/Output/Transcode:** SRT (Caller, Listener, Rendezvous), ZIXI (Pull, Feeder)
Server Address:	IP address of the MULTIDYNE **FASP Server**.	**Input:** FASP Client
FASP Port:	The port used for data transfer in the FASP connection.	**Input:** FASP Client

Target Rate:	Overall stream rate, matches the transport rate of the encoder profile.	**Input:** FASP Client
Remote ID / Password:	These fields represent the **Remote ID** and **Password** of a Zixi broadcaster, which may be optional depending on the desired stream.	**Input:** ZIXI Pull **Output/Transcode:** ZIXI Feeder (password only)
Stream ID:	This field represents the stream ID of a Zixi broadcaster and an **RTMP Server**. Clicking the **Regenerate** button will create a new RTMP **Stream ID** and replace the content in the field if there is any.	**Input:** ZIXI Pull, RTMP Server **Output/Transcode:** ZIXI Feeder
FEC:	Turns on Forward Error Correction, for use with Zixi Broadcasters to retransmit dropped or corrupted packets.	**Input:** ZIXI (Pull, Accept)
FEC Overhead:	Sets the percentage of additional bandwidth to be allotted for FEC use.	**Input:** ZIXI (Pull, Accept) **Output/Transcode:** ZIXI Feeder
FEC Block:	The maximum time given to the FEC to correct an issue. This should not be higher than half of the Latency setting.	**Input:** ZIXI (Pull, Accept) **Output/Transcode:** ZIXI Feeder
FEC Aware:	Allocates the FEC packets based on content.	**Input:** ZIXI (Pull, Accept) **Output/Transcode:** ZIXI Feeder
URL:	This field represents the URL address of a stream. The **RTMP** fields is tied to its respective Protocol, while the **Custom** field allows users to enter an unspecified stream address that the decoder will parse. The **HLS** second field can be changed to a personalized **Stream ID**. Copy the whole path to receive the stream.	**Input:** RTMP (Client, Server), Custom **Transcode:** HLS
TTL:	**Time to Live** number of hops between devices, default value is 64.	**Output/Transcode:** UDP/RTP, SRT (Caller, Listener, Rendezvous)
Traffic Shaping:	Buffers the output data to ensure constant intervals between packets.	**Output/Transcode:** UDP/RTP
Bandwidth Overhead:	Percentage used for SRT error correction and retransmitting packets.	**Output/Transcode:** SRT (Caller, Listener, Rendezvous)
Input BW:	The current input bitrate of the SRT stream to send. Used by Bandwidth Overhead to determine the percentage of extra bandwidth to be allowed for error correction.	**Output/Transcode:** SRT (Caller, Listener, Rendezvous)

Max BW:	Sets a hard target value for the total bandwidth limit, comprised of bandwidth used for the A/V bitrate as well as any error correction and SRT overhead.	**Output/Transcode:** SRT (Caller, Listener, Rendezvous)
ToS:	**Type of Service** is the payload type for prioritizing traffic through routers.	**Output/Transcode:** SRT (Caller, Listener, Rendezvous)
Max Bitrate:	The rate of the transport output should be set equal or greater than the total output of the device. Recommended to set 20% higher.	**Output/Transcode:** ZIXI Feeder
Min Bitrate:	Setting a value enables Adaptive Bitrate.	**Output/Transcode:** ZIXI Feeder
Output Location:	List of available directories.	**Output/Transcode:** File
Filename:	The name of the file which will be created.	**Output/Transcode:** File

INPUTS

UDP/RTP and TCP

From the Input, **UDP/RTP** and **TCP** can be used for receiving a unicast or multicast stream. For a unicast, the **Address** entered is not important; only the **Interface** and **Port** must be correct according to the outputting device. For a multicast, all 3 fields must be correct according to the outputting device.

Protocol	Interface	Address	Port	
UDP / RTP ▾	Gig 2 ▾	239.0.30.30	2000	Apply

Protocol	Interface	Address	Port	
TCP ▾	Gig 2 ▾	239.0.30.30	2000	Apply

Configuration Information	
Interface:	Choose an active Network Interface, displayed on the Network Tab (**UP**) TCP requires GigE port 2.
Address:	Working with UDP and RTP, typically leading with 239 is for multicasting. For unicasting enter the localhost address.
Port:	Match the port that the stream is being output to. Accepts a number between 1000-16000.

SRT (Caller and Rendezvous)

From the Input, **SRT Caller** requires the **Address** and **Port** of the device outputting at **SRT Listener**. SRT Rendezvous requires the the **Address** and **Port** of the device outputting at **SRT Rendezvous**.

Configuration Information	
Address:	The address of the source stream device.
Port:	Match the port the stream is being output to.
Encryption and Password:	Set an encryption type on a password.
Timeout:	Amount of time to wait before connection experiences timeout. When left blank, factory default is 3 seconds.
Latency:	When left blank, factory default value is 3000 milliseconds. This value is typically set by the source stream device, but can be set here. The larger value of the two will be used during the connection.

SRT (Listener)

From the Input, **SRT Listener** provides the **Port** which the device outputting at **SRT Caller** will send to.

Configuration Information	
Port:	Match the port the stream is being output to.
Encryption and Password:	Set an encryption type on a password.
Latency:	When left blank, factory default value is 3000 milliseconds. This value is typically set by the source stream device, but can be set here. The larger value of the two will be used during the connection.

FASP (Client)

From the Input, **FASP (Client)** specifies the **Server Address** providing the unicast stream, the **FASP Port** at which the server should send data, and the **Target Rate** in bps at which to receive the stream.

Protocol	Server Address	FASP Port	Target Rate	
FASP (Client) ▼				Apply

Configuration Information	
Server Address:	The address of the source streaming device.
FASP Port:	The port used for data transfer in the connection. Default is 33001.
Target Rate:	Set the target bitrate to receive in bits per second (bps).

ZIXI (Pull)

From the Input, **ZIXI (Pull)** specifies the details of the Zixi Broadcaster which is outputting the stream.

Protocol	Address	Port	
ZIXI (Pull) ▼			Apply

Stream ID	Remote ID	Password	Latency(ms)	FEC
				True ▼

FEC Overhead	FEC Block	FEC Aware	Encryption Type
15 %	30	False ▼	None ▼

Configuration Information	
Remote ID and Password:	These are specified by the Zixi Broadcaster which is providing the output stream.
Address:	The address of your Zixi Broadcaster which is providing the output stream.
Port:	Match the port the stream is being output to. The default output port for a Zixi Broadcaster is 2077.
Stream ID:	The Stream ID is specified on the Zixi Broadcaster.
Latency:	Increasing latency will improve stream quality in poor network scenarios. When left blank, the default value is 3000 milliseconds.
FEC:	Turns on FEC, or Forward Error Correction, for use in retransmitting dropped or corrupted packets. The default setting is off, or False. Setting FEC to true will expose the additional fields FEC Overhead, FEC Block, and FEC Aware.
FEC Overhead:	Sets the percentage of additional bandwidth to be allotted for FEC use. The default value is 15%.
FEC Block:	The maximum time given to the FEC to correct an issue. This should not be higher than half of the Latency setting.
FEC Aware:	Allocates the FEC packets based on content.
Encryption Type:	Sets an encryption type on a password key. The encryption type must match the type used by the Zixi Broadcaster.

ZIXI (Accept)

From Input, **ZIXI (Accept)** only needs to specify a port for the Zixi Broadcaster to use for sending the stream.

Protocol

ZIXI (Accept)

Port

Apply

FEC

True

FEC Overhead

30   %

FEC Block

50

FEC Aware

False

Encryption Type

None

Configuration Information	
Port:	The port at which the Zixi Broadcaster will send the stream.
FEC:	Turns on FEC, or Forward Error Correction, for use in retransmitting dropped or corrupted packets. The default setting is off, or False. Setting FEC to true will expose the additional fields FEC Overhead, FEC Block, and FEC Aware.
FEC Overhead:	Sets the percentage of additional bandwidth to be allotted for FEC use. The default value is 15%.
FEC Block:	The maximum time given to the FEC to correct an issue. This should not be higher than half of the Latency setting.
FEC Aware:	Allocates the FEC packets based on content.
Encryption Type:	Sets an encryption type on a password key. The encryption type must match the type used by the Zixi Broadcaster.

RTMP (Client)

From Input, **RTMP (Client)** requires only the full address of the RTMP stream provided by the RTMP server.

Protocol

RTMP (Client)

URL

Apply

Configuration Information	
URL:	Provide the full address for receiving the RTMP stream.

RTMP (Server)

From Input, **RTMP (Server)** puts the device into a listening mode. The information shown is what will be used by the outputting device which is acting as the RTMP client.

Configuration Information	
URL:	This is the local address of your device. If the RTMP Client is located on the public internet, the RTMP client should instead argue your public address and the port where port 1935 has been forwarded to.
Stream ID:	This ID can be manually set or randomly generated. For a new key, click Regenerate. Otherwise, enter your desired key. It will be applied upon clicking Apply.
Copy Path:	Clicking this button will bring up the full path to be used by your RTMP client for connecting to your device. It is the concatenation of URL and Stream ID. This is for easier copying of the path to enter on your RTMP client.

Custom

Use to stream HLS, RTSP streams.

Protocol	URL	
Custom ▼		Apply

Configuration Information	
URL:	Type the HLS URL or RTSP stream address for the source. HLS Example: https://devstreaming-cdn.apple.com/videos/streaming/examples/img_bipbop_adv_example_ts/v6/prog_index.m3u8

OUTPUTS

UDP

User datagram protocol used to send data over a network. Supports unicast and multicast.

Protocol	Interface	Address	Port	TTL	Bitrate	Traffic Shaping	
UDP	Gig 2	239.0.58.58	2000	9		☐	Apply

Configuration Information	
Interface:	Select the GigE interface you wish to stream from.
Address:	For multicast, enter the streams multicast address. Example: 239.0.24.24:2000. For unicast, enter the IP address of the device you are streaming to. Example: 192.168.2.7:2000 if the interface is MULTIDYNE default IP address for Gige2. The device streaming to the MULTIDYNE decoder must be configured to stream to 192.168.2.7 on port 2000. Unicasts are much friendlier to corporate networks. Multicast can take corporate networks down if they are not configured to support Multicast and IGMP.
Port:	The port the stream is carried on.

TTL:	Time To Live. This indicates the number of switches a packet can traverse before the packet is considered stale and destroyed.
Bitrate:	Used in conjunction with Traffic Shaping, this specifies the bitrate goal for the traffic shaping algorithm in bits per second (bps). This must be matched as closely as possible to the input bitrate. Use the information from the source streaming device or the bitrate graph in the upper right of the UI to get an approximation of the input rate.
Traffic Shaping:	Turns on the traffic rate shaping for your output stream. This is more resource-intensive, but produces a smooth bitrate stream.

RTP

RTP protocol adds packet numbering headers to UDP. The numbered headers enables the decoder to reorder the packets before decoding the stream. RTP is considered more reliable than UDP.

Protocol	Interface	Address	Port	Bitrate	Traffic Shaping	
RTP	Gig 2	239.0.58.58	2000		☐	Apply

Configuration Information	
Interface:	Select the GigE interface you wish to stream from.
Address:	For multicast, enter the streams multicast address. Example: 239.0.24.24:2000. For unicast, enter the IP address of the port you are streaming too. Example: 192.168.2.7:2000 if the interface is MULTIDYNE default IP address for Gige2. The device streaming to the MULTIDYNE decoder must be configured to stream to 192.168.2.7 on port 2000. Unicasts are much friendlier to corporate networks. Multicast can take corporate networks down if they are not configured to support Multicast and IGMP.
Port:	The port the stream is carried on.
Bitrate:	Used in conjunction with Traffic Shaping, this specifies the bitrate goal for the traffic shaping algorithm in bits per second (bps). This must be matched as closely as possible to the input bitrate. Use the information

	from the source streaming device or the bitrate graph in the upper right of the UI to get an approximation of the input rate.
Traffic Shaping:	Turns on the traffic rate shaping for your output stream. This is more resource-intensive, but produces a smooth bitrate stream.

SRT (Caller and Rendezvous)

From the Output, **SRT Caller** requires the **Address** and **Port** of the device receiving at **SRT Listener**. SRT Rendezvous requires the the **Address** and the mutually agreed on **Port** of the device rendezvousing at **SRT Rendezvous**.

Configuration Information	
Address:	SRT is a single connection protocol. The address is the IP address of the destination device.
Port:	Match the port the stream is being output to.
Encryption Type and Password:	Set an encryption type on a password. The password can be 10-79 characters.
Bandwidth Overhead:	SRT Bandwidth Overhead is calculated as a percentage of the A/V bit rate, such that the sum of the two represents a threshold bit rate,

	which is the maximum bandwidth the SRT stream is expected to use. The default is 25%, the maximum is 50%. This represents the maximum allotted bandwidth which can be used for stream correction on noisy networks.
Input BW:	The current input bitrate of the SRT stream to send. When transcoding, this is the configured video bitrate. This is used by Bandwidth Overhead to determine the percentage of extra bandwidth to be allowed for error correction. If left empty, the input bitrate will be evaluated internally by the SRT library. If using Max BW to set a target bandwidth, leave empty.
Max BW:	An alternative method for setting a hard target value for the total bandwidth limit, comprised of bandwidth used for the A/V bitrate as well as any error correction and SRT overhead. When using this, Input BW and Bandwidth Overhead can be left empty. Setting a -1 here represents 'infinite', and allows up to the maximum SRT bandwidth (roughly 30mbps).
MTU:	Maximum packet size of the UDP-based packet. Maximum possible is 1500, default is 1496.
TTL:	Time to Live, the maximum number of hops a packet can make between devices before it is considered stale and destroyed.
ToS:	The Type of Service field in the IPv4 header, which sets a priority value on the packet.
Timeout:	Amount of time to wait before connection experiences timeout. When left blank, factory default is 3 seconds.
Latency:	A fixed value (from 20 to 8000 ms) representing the maximum buffer size available for managing SRT packets. When left blank, factory default value is 3000 milliseconds. The larger value of the two (set by source and destination) will be used during the connection.

SRT Listener

From the Output, **SRT Listener** specifies the **Port** which the device will listen at for an **SRT Caller** connection.

Protocol	Port	Timeout	Latency(ms)	
SRT (Listener) ▼				Apply

Bandwidth Overhead	MTU	TTL	ToS
25 %	1496	64	0xB8

Input BW (bytes/sec)	Max BW (bytes/sec)	Encryption Type	Password
		AES 128 ▼	

Configuration Information	
Port:	The port value where the transcoder will listen for incoming SRT Caller connections.
Encryption Type and Password:	Set an encryption type on a password. The password can be 10-79 characters.
Timeout:	Amount of time to wait before connection experiences timeout. When left blank, factory default is 10 seconds.
Latency:	A fixed value (from 20 to 8000 ms) representing the maximum buffer size available for managing SRT packets. When left blank, factory default value is 3000 milliseconds. The larger value of the two (set by source and destination) will be used during the connection.
Bandwidth Overhead:	SRT Bandwidth Overhead is calculated as a percentage of the A/V bit rate, such that the sum of the two represents a threshold bit rate, which is the maximum bandwidth the SRT stream is expected to use. The default is 25%, the maximum is 50%. This represents the maximum allotted bandwidth which can be used for stream correction on noisy networks.
Input BW:	The current input bitrate of the SRT stream to send. When transcoding, this is the configured video bitrate. This is used by Bandwidth Overhead to determine the percentage of extra bandwidth to be allowed for error correction. If left empty, the input bitrate will be evaluated internally by the SRT library. If using Max BW to set a target bandwidth, leave empty.
Max BW:	An alternative method for setting a hard target value for the total bandwidth limit, comprised of bandwidth used for the A/V bitrate as

	well as any error correction and SRT overhead. When using this, Input BW and Bandwidth Overhead can be left empty. Setting a -1 here represents 'infinite', and allows up to the maximum SRT bandwidth (roughly 30mbps).
MTU:	Maximum packet size of the UDP-based packet. Maximum possible is 1500, default is 1496.
TTL:	Time to Live, the maximum number of hops a packet can make between devices before it is considered stale and destroyed.
ToS:	The Type of Service field in the IPv4 header, which sets a priority value on the packet.

ZIXI Feeder

From the Output, **ZIXI Feeder** requires the **Address** and **Port** of the Zixi Broadcaster which will receive the stream as its input.

Protocol	Address			Port	
ZIXI (Feeder) ▾					Apply

Stream ID	Password	Latency(ms)	Max Bitrate(bps)	Min Bitrate(bps) ☐

FEC Overhead	FEC Block	FEC Aware
15 %	30	False ▾

Encryption Type

None ▾

Configuration Information

Address:	The address of the Zixi Broadcaster, which may be an IP address or a domain name (proper DNS settings must be set on your device).
Port:	The input port of the Zixi Broadcaster. Default is 2088.
Stream ID:	The ID used by the Zixi Broadcaster to identify the stream. This must be set up on the Broadcaster before the Zixi Feeder is started.
Password:	The password specified on the Zixi Broadcaster.
Latency:	Increasing latency will improve stream quality in poor network scenarios. When left blank, the default value is 3000 milliseconds.
Max Bitrate:	Set the maximum bitrate to be sent to the Zixi Broadcaster. 20% higher than the configured video bitrate is recommended.
Min Bitrate:	Setting a minimum bitrate is used when an adaptive bitrate output stream is desired. Checking the box will enable the field to enter a bitrate.
FEC Overhead:	Sets the percentage of additional bandwidth to be allotted for FEC use (Forward Error Correction, for retransmitting dropped or corrupted packets). The default value is 15%.
FEC Block:	The maximum time given to the FEC to correct an issue. This should not be higher than half of the Latency setting.
FEC Aware:	Allocates the FEC packets based on content.
Encryption Type and Key:	Set an encryption type and passphrase. Decrypting on the Zixi Broadcaster is only necessary for using the Zixi Broadcaster to analyze the stream. Otherwise, not decrypting it will carry the encryption over its use as an output stream.

File

Use this feature to record the incoming stream to disk.

Protocol	Output Location Filename	
FILE ▾	/disk/media/ ▾	Apply

Configuration Information	
Output Location:	Disk mount where the file will be recorded.
Filename:	User defined file name.

TCP

From the Output, **TCP** requires the **Port** where the device will listen for incoming connections.

Protocol   Port

TCP ▼   [   ]   Apply

Configuration Information	
Port:	Port value where the device will listen for connections.

TRANSCODE

UDP and RTP

User datagram protocol used to send data over a network. RTP protocol adds packet numbering headers to UDP. The numbered headers enables the decoder to reorder the packets before decoding the stream. RTP is considered more reliable than UDP. Supports unicast and multicast.

Transcode 1 ⓘ

Protocol	Interface	Address	Port	TTL	Traffic Shaping	
UDP ▼	Gig 2 ▼			9	☐	Apply
RTP ▼	Gig 2 ▼			9	☐	Apply

Configuration Information

Interface:	Select the GigE interface you wish to stream from.
Address:	For multicast, enter the streams multicast address. Example: 239.0.24.24:2000. For unicast, enter the IP address of the device you are streaming to. Example: 192.168.2.7:2000 if the interface is MULTIDYNE default IP address for Gige2. The device streaming to the MULTIDYNE decoder must be configured to stream to 192.168.2.7 on port 2000. Unicasts are much friendlier to corporate networks. Multicast can take corporate networks down if they are not configured to support Multicast and IGMP.
Port:	The port the stream is carried on.
TTL:	Time To Live, which specifies the number of hops a packet can make between devices before it is considered stale and destroyed..
Bitrate:	Used in conjunction with Traffic Shaping, this specifies the bitrate goal for the traffic shaping algorithm in bits per second (bps). This must be matched as closely as possible to the input bitrate. Use the information from the source streaming device or the bitrate graph in the upper right of the UI to get an approximation of the input rate.
Traffic Shaping:	Turns on the traffic rate shaping for your output stream. This is more resource-intensive, but produces a smooth bitrate stream.

SRT (Caller and Rendezvous)

From the Output, **SRT Caller** requires the **Address** and **Port** of the device receiving at **SRT Listener**. SRT Rendezvous requires the the **Address** and the mutually agreed on **Port** of the device rendezvousing at **SRT Rendezvous**.

Protocol	Address					Port	
SRT (Caller) ▼							Apply
Bandwidth Overhead	**MTU**	**TTL**	**ToS**	**Timeout**	**Latency(ms)**		
25 %	1496	64	0xB8				
Input BW (bytes/sec)	**Max BW (bytes/sec)**	**Encryption Type**	**Password**				
		AES 128 ▼					

Protocol	Address					Port	
SRT (Rendezvous) ▼							Apply
Bandwidth Overhead	**MTU**	**TTL**		**ToS**	**Timeout**	**Latency(ms)**	
25 %	1496	64		0xB8			
Input BW (bytes/sec)	**Max BW (bytes/sec)**	**Encryption Type**		**Password**			
		AES 128 ▼					

Configuration Information	
Address:	SRT is a single connection protocol. The address is the IP address of the destination device.
Port:	Match the port the stream is being output to.
Encryption Type and Password:	Set an encryption type on a password. The password can be 10-79 characters.
Bandwidth Overhead:	SRT Bandwidth Overhead is calculated as a percentage of the A/V bit rate, such that the sum of the two represents a threshold bit rate, which is the maximum bandwidth the SRT stream is expected to use. The default is 25%, the maximum is 50%. This represents the maximum allotted bandwidth which can be used for stream correction on noisy networks.
Input BW:	The current input bitrate of the SRT stream to send. When transcoding, this is the configured video bitrate. This is used by Bandwidth Overhead to determine the percentage of extra bandwidth to be allowed for error correction. If left empty, the input bitrate will be evaluated internally by the SRT library. If using Max BW to set a target bandwidth, leave empty.
Max BW:	An alternative method for setting a hard target value for the total bandwidth limit, comprised of bandwidth used for the A/V bitrate as well as any error correction and SRT overhead. When using this, Input BW and Bandwidth Overhead can be left empty. Setting a -1 here represents 'infinite', and allows up to the maximum SRT bandwidth (roughly 30mbps).
MTU:	Maximum packet size of the UDP-based packet. Maximum possible is 1500, default is 1496.

TTL:	Time to Live, the maximum number of hops a packet can make between devices before it is considered stale and destroyed.
ToS:	The Type of Service field in the IPv4 header, which sets a priority value on the packet.
Timeout:	Amount of time to wait before connection experiences timeout. When left blank, factory default is 10 seconds.
Latency:	A fixed value (from 20 to 8000 ms) representing the maximum buffer size available for managing SRT packets. When left blank, factory default value is 3000 milliseconds. The larger value of the two (set by source and destination) will be used during the connection.

SRT Listener

From the Output, **SRT Listener** specifies the **Port** which the device will listen at for an **SRT Caller** connection.

Protocol	Port	Timeout	Latency(ms)	
SRT (Listener) ▼				Apply

Bandwidth Overhead	MTU	TTL	ToS
25 %	1496	64	0xB8

Input BW (bytes/sec)	Max BW (bytes/sec)	Encryption Type	Password
		AES 128 ▼	

Configuration Information	
Port:	Match the port the stream is being output to.
Encryption Type and Password:	Set an encryption type on a password. The password can be 10-79 characters.
Timeout:	Amount of time to wait before connection experiences timeout. When left blank, factory default is 10 seconds.

Latency:	A fixed value (from 20 to 8000 ms) representing the maximum buffer size available for managing SRT packets. When left blank, factory default value is 3000 milliseconds. The larger value of the two (set by source and destination) will be used during the connection.
Bandwidth Overhead:	SRT Bandwidth Overhead is calculated as a percentage of the A/V bit rate, such that the sum of the two represents a threshold bit rate, which is the maximum bandwidth the SRT stream is expected to use. The default is 25%, the maximum is 50%. This represents the maximum allotted bandwidth which can be used for stream correction on noisy networks.
Input BW:	The current input bitrate of the SRT stream to send. When transcoding, this is the configured video bitrate. This is used by Bandwidth Overhead to determine the percentage of extra bandwidth to be allowed for error correction. If left empty, the input bitrate will be evaluated internally by the SRT library. If using Max BW to set a target bandwidth, leave empty.
Max BW:	An alternative method for setting a hard target value for the total bandwidth limit, comprised of bandwidth used for the A/V bitrate as well as any error correction and SRT overhead. When using this, Input BW and Bandwidth Overhead can be left empty. Setting a -1 here represents 'infinite', and allows up to the maximum SRT bandwidth (roughly 30mbps).
MTU:	Maximum packet size of the UDP-based packet. Maximum possible is 1500, default is 1496.
TTL:	Time to Live, the maximum number of hops a packet can make between devices before it is considered stale and destroyed.
ToS:	The Type of Service field in the IPv4 header, which sets a priority value on the packet.

FASP Server

For Transcoding, no additional options need to be set when using **FASP Server**. The **FASP Client** will specify the address of the device to receive from and the target bitrate. No port or interface need to be specified as it is a one-to-one device connection.

Protocol

FASP (Server) ▼   Apply

ZIXI Feeder

ZIXI Feeder requires the **Address** and **Port** of the Zixi Broadcaster which will receive the stream as its input.

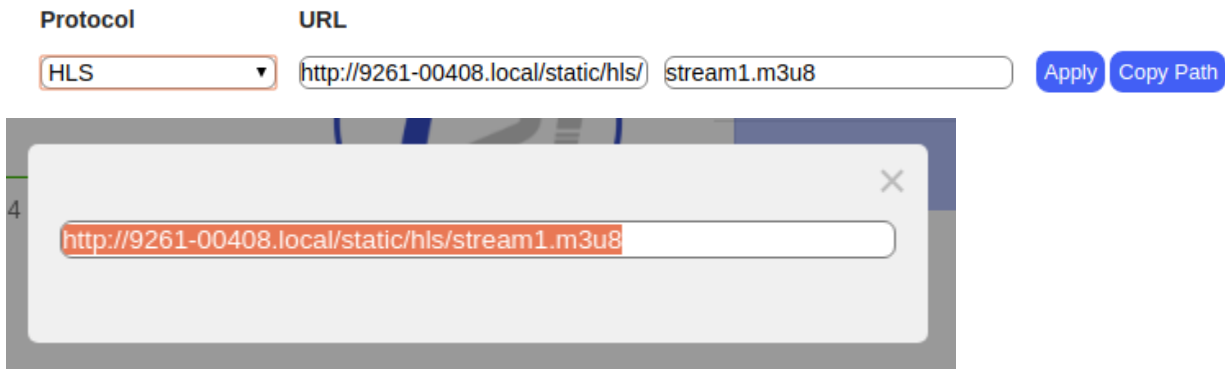|					
Protocol	**Address**				**Port**
ZIXI (Feeder) ▼						Apply
Stream ID	**Password**	**Latency(ms)**	**Max Bitrate(bps)**	**Min Bitrate(bps)** ☐	
FEC Overhead	**FEC Block**	**FEC Aware**			
15 %	30	False ▼			
Encryption Type					
None ▼					

Configuration Information	
Address:	The address of the Zixi Broadcaster, which may be an IP address or a domain name (proper DNS settings must be set on your device).
Port:	The input port of the Zixi Broadcaster. Default is 2088.
Stream ID:	The ID used by the Zixi Broadcaster to identify the stream. This must be set up on the Broadcaster before the Zixi Feeder is started.
Password:	The password specified on the Zixi Broadcaster.
Latency:	Increasing latency will improve stream quality in poor network scenarios. When left blank, the default value is 3000 milliseconds.
Max Bitrate:	Set the maximum bitrate to be sent to the Zixi Broadcaster.
Min Bitrate:	Setting a minimum bitrate is used an adaptive bitrate output stream is desired. Checking the box will enable the field to enter a bitrate.

FEC Overhead:	Sets the percentage of additional bandwidth to be allotted for FEC use (Forward Error Correction, for retransmitting dropped or corrupted packets). The default value is 15%.
FEC Block:	The maximum time given to the FEC to correct an issue. This should not be higher than half of the Latency setting.
FEC Aware:	Allocates the FEC packets based on content.

HLS

HLS generates .ts file segments and a manifest file, and the URL signifies the name of the device and the location at which the manifest file can be accessed.

Configuration Information	
URL:	The first field cannot be edited, but shows the URL at which the stream can be accessed; this is a URL to your device, and as such, this can be an IP address (public or local), or a name such as the device's Zero Conf interface address shown above. The second field can be edited to a personalized stream ID.
Copy Path:	For easy access to entire path for copying purposes click this button.

RTMP Client

Typically used to livestream to sites like Youtube or Twitch, this can also be used to output to other MULTIDYNE devices. The URL is manufactured with a key unique to your account provided by the site.
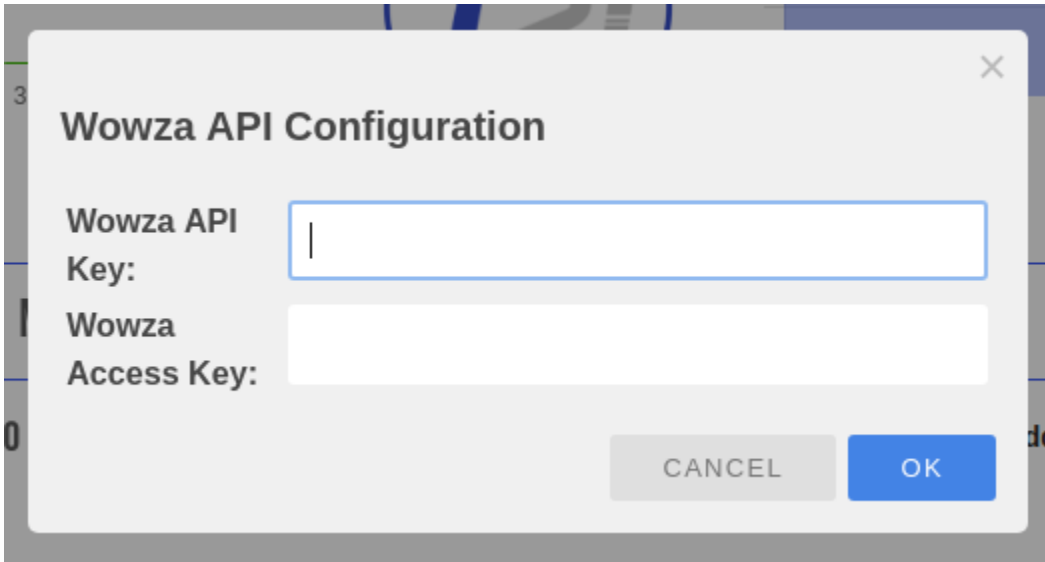

Configuration Information	
URL:	This is the full path to the RTMP server. For Wowza, it is the address of the Wowza server, provided by Wowza. In the second field, you will provide the key specific to your account as given by the site.
Advanced Options	
Wowza API Key:	Provided by the Wowza user credentials on the Wowza server.
Wowza Access Key:	Provided by the Wowza user credentials on the Wowza server.

File

Use this feature to record the transcoded stream to disk.

Configuration Information	
Output Location:	Disk mount where the file will be recorded.
Filename:	User defined file name.