Enhanced RTMP (V2)

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Document Status

Author: Slavik Lozben (Veovera Software Organization)(VSO)
Contributors: Adobe, Google, Twitch, Jean-Baptiste Kempf (FFmpeg, VideoLAN), pkv (OBS), Dennis Sädtler (OBS), Xavier Hallade (Intel Corporation), Luxoft, SplitmediaLabs Limited (XSplit), Craig Barberich (VSO), Michael Thornburgh

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Documentation Versioning

Overview

This section outlines our standardized approach for versioning our specification documentation. Effective versioning ensures consistency, enables users to identify the latest version easily, and facilitates collaboration among team members.

File Naming Convention

We name the documentation files with a clear identifier and the major version number.

Example:
enhanced-rtmp-v2.pdf

Version Information Inside the Document

We include a dedicated section or metadata within each document to specify the version details which includes the major version number, date, and stage of development (alpha/beta/release).

Example:
Status: v2-2024-02-26-a1

Calendar Versioning Format Description

The format for versioning documents is structured as follows:

- **v#-yyy-mm-dd-[a|b|r]#:**
  - **v#:** Major version number for tracking the progression of the E-RTMP development.
  - **yyyy-mm-dd:** Date when the document was updated.
  - **[a|b|r]:** Suffix to distinguish between the alpha, beta, and release stage.
  - **#:** Minor version number for a particular date. Increments for multiple versions on the same date.

This format provides a comprehensive overview of each version's status and chronological order, facilitating effective tracking and management of the E-RTMP specification development.

Alpha Version Disclaimer for Enhanced RTMP v*

This document details an 'alpha version' of the Enhanced Real-Time Messaging Protocol (a.k.a., E-RTMP) specification, version "*". As we continue to refine and enhance the protocol, we remain open to implementing necessary updates based on user feedback and further testing. While there is a possibility of introducing breaking changes during the alpha stage, we are committed to maintaining the integrity of the General Availability (GA) versions and strive to ensure they remain free from breaking changes.

We encourage developers, implementers, and stakeholders to actively participate in this development phase. Your feedback, whether it be bug reports, feature suggestions, or usability improvements, is invaluable and can be submitted via new issues in our GitHub repository at <https://github.com/veovera/enhanced-rtmp>. We are committed to transparently communicating updates and changes, ensuring that all stakeholders are informed and involved.

Engaging with the alpha version provides a unique opportunity to influence the final specifications of E-RTMP version "*". We value your input and look forward to collaborating with you on this exciting journey towards developing a more robust and efficient protocol.

Usage License

Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here. Definitions below are reproduced from [RFC2119].

- **MUST** - This word, or the terms "REQUIRED" or "SHALL", means that the definition is an absolute requirement of the specification.

- **MUST NOT** - This phrase, or the phrase "SHALL NOT", means that the definition is an absolute prohibition of the specification.
- **SHOULD** - This word, or the adjective "RECOMMENDED", means that there may exist valid reasons in particular circumstances to ignore a particular item, but the full implications must be understood and carefully weighed before choosing a different course.
- **SHOULD NOT** - This phrase, or the phrase "NOT RECOMMENDED", means that there may exist valid reasons in particular circumstances when the particular behavior is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behavior described with this label.
- **MAY** - This word, or the adjective "OPTIONAL", means that an item is truly optional. One vendor may choose to include the item because a particular marketplace requires it or because the vendor feels that it enhances the product while another vendor may omit the same item. An implementation which does not include a particular option MUST be prepared to interoperate with another implementation which does include the option, though perhaps with reduced functionality. In the same vein an implementation which does include a particular option MUST be prepared to interoperate with another implementation which does not include the option (except, of course, for the feature the option provides.)

Additionally we add the keyword [DEPRECATED] to the set of keywords above.

- **DEPRECATED** - This word means a discouragement of use of some terminology, feature, design, or practice, typically because it has been superseded or is no longer considered efficient or safe, without completely removing it or prohibiting its use. Typically, deprecated materials are not completely removed to ensure legacy compatibility or back-up practice in case new methods are not functional in an odd scenario. It can also imply that a feature, design, or practice will be removed or discontinued entirely in the future.

Abstract

In the rapidly evolving media streaming landscape, there is a pressing need to update legacy protocols to align with modern technological standards. The Real-Time Messaging Protocol [RTMP] and Flash Video [FLV] file format, introduced in 2002, have been pivotal and continue to be vital especially in live broadcasting. Despite RTMP widespread use, it has shown signs of aging, particularly in the lack of support for contemporary video codecs (VP8, VP9, HEVC, AV1) and audio codecs (Opus, FLAC, AC-3, E-AC-3). Recognizing this, Veovera Software Organization (VSO), in collaboration with industry giants like Adobe, YouTube, and Twitch, and other key stakeholders, has embarked on a mission to rejuvenate RTMP, ensuring it meets the demands of contemporary streaming needs.

This document details the comprehensive enhancements made to the RTMP and FLV specifications, aimed at revitalizing the technology for current and future media demands. Our strategic approach prioritizes innovation while maintaining backward compatibility, thereby augmenting RTMP's utility without undermining existing infrastructures. Some of the key advancements include:

- **Advanced Audio Codecs:** Integration of codecs like AC-3, E-AC-3, Opus, and FLAC to meet diverse audio quality and compression needs, ensuring compatibility with modern systems.
- **Multichannel Audio Configurations:** Support for multichannel audio to enhance auditory experiences without compromising existing setups.
- **Advanced Video Codecs:** Introduction of codecs such as VP8, VP9, HEVC, and AV1 with HDR support to meet modern display and content standards.
- **Video Metadata:** Expansion of VideoPacketType.Metadata to support a broader range of video metadata types.

- **FourCC Signaling:** Inclusion of FourCC signaling for advanced codecs mentioned above, as well as for legacy codecs such as AVC, AAC, and MP3.
- **Multitrack Capabilities:** New audio and video multitrack capabilities for concurrent management and processing of multiple media streams, enhancing media experiences.
- **Reconnect Request Feature:** A new Reconnect Request feature improves connection stability and resilience.

These strategic enhancements position RTMP as a robust, future-proof standard in the streaming technology arena. Veovera is committed to open collaboration and values community input. We encourage participation in the ongoing development process through our GitHub repository, where you can access detailed documentation, contribute to the project, and share insights, fostering a vibrant ecosystem around enhanced E-RTMP.

Introduction

This document describes enhancements to legacy [RTMP] and legacy [FLV], introducing support for new media codecs, HDR capability, and more. A primary objective is to ensure these enhancements do not introduce breaking changes for established clients or the content they stream. As such, legacy RTMP and legacy FLV specifications remain integral to the RTMP ecosystem. While this updated specification aims to minimize redundancy with previous versions, when combined with previous-generation documentation, it provides a comprehensive overview of the RTMP solution. We've drawn from several legacy references, which are as follows:

- Adobe Legacy [RTMP] Specification
- Adobe Legacy [FLV] Specification
- Additional [LEGACY] Specifications

Conventions

This document employs certain conventions to convey particular meanings and requirements. The following section outlines the notation, terminology, and symbols used throughout to ensure clarity and consistency. These conventions provide insight into the ethos of how the E-RTMP specification has been crafted and should be interpreted.

- **Enhanced RTMP:** refers to a series of improvements made to the legacy Real-Time Messaging Protocol (RTMP), originally developed by Adobe. It's important to note that **enhanced RTMP** is not a brand name but a term used to distinguish this advanced version from the legacy [RTMP] specification. Endorsed by Adobe and widely adopted across the industry, enhanced RTMP serves as the current standard for RTMP solutions. This updated protocol includes various enhancements to both RTMP and the [FLV] format. Please be aware that the term **enhanced RTMP** (a.k.a., **E-RTMP**) signifies ongoing updates to RTMP and FLV, and does not pertain to any specific iteration or release.
- Pseudocode: Pseudocode has been provided to convey logic on how to interpret the E-RTMP or FLV binary format. The code style imitates a cross between TypeScript and C. The pseudocode was written in TypeScript and validated using VSCode to ensure correct syntax and catch any minor typographical errors. Below are some further explanations:

    - Enumerations are used to define valid values
    - Pseudo variables are named in a self-descriptive manner. For instance:

      videoCommand = UI8 as VideoCommand

      The line above indicates that an unsigned 8-bit value is read from the bitstream. The legal values correspond to the enumerations within the **VideoCommand** set, and the pseudo variable **videoCommand** now holds that value.
    - The pseudocode is written from the point of view of reading (a.k.a., parsing) the bitstream. If you are writing the bitstream, you can swap source with destination variables.
    - E-RTMP typically employs camelCase naming conventions for variables. In contrast, the naming convention for legacy RTMP specification is usually preserved as is.
    - Handshake and Enhancing NetConnection **connect** command: The E-RTMP specification generally prioritizes the client's perspective over that of the server. To shift this focus and view the interaction from the server's side, the server should echo back certain enhancement information.

      When the client informs the server of the enhancements it supports via the **connect** command, the server processes this command and responds using the same transaction ID. The server's response string will be one of the following: **_result**, **_error**, or a specific method name. A command string of **_result** or **_error** indicates a response rather than a new command.

      During this response, the server will include an object containing specific properties as one of the arguments to **_result**. It is at this point that the server should indicate its support for E-RTMP features. Specifically, the server should denote its capabilities through attributes such as **videoFourCcInfoMap**, **capsEx**, and other defined properties.
    - The ethos of this pseudocode is to provide a high-level overview of the data structures and operations taking place on the wire. While it accurately represents the bytes being transmitted, it's important to note that the logic is not exhaustive. Specifically, this pseudocode does not cover all possible cases, nor does it always include items such as initialization logic, looping logic or error-handling mechanisms. It serves as a foundational guide that can be implemented in various ways, depending on specific needs and constraints.

- **Unrecognized value**: If a value in the bitstream is not understood, the logic must fail gracefully in a manner appropriate for the implementation.
- **Table naming**: Each table in the document is named according to the specific content or subject it is describing.
- **Bitstream optimization**: One of the guiding principles of E-RTMP is to optimize the number of bytes transmitted over the wire. While minimizing payload overhead is a priority, it is sometimes more important to simplify the logic or enhance extensibility. For example, although more optimal methods for creating a codec ID than using FOURCC may exist, such approaches could render the enhancement non-standard and more challenging to extend and maintain in the future.
- **Capitalization rules**: Another guiding principle in the E-RTMP is the standardization of capitalization for types. The original documentation capitalized types such as Number, String, and Boolean, and even included various other spellings. The E-RTMP adopts lowercase spelling for terms, such as number, string, and boolean. This change emphasizes that these types are simple, not objects.
- **ECMA Array vs Object:** In the world of AMF (Action Message Format), both ECMA Array and Object are used to store collections of properties. A property is simply a pairing of a name with a value. In enhanced RTMP, the term **Object** is specifically used to indicate the Object Type. In the past, people have sometimes used **ECMA Array** and **Object** as if they were the same thing. However, for better

coding practices, it's recommended to use **Object** when you're creating AMF data. When you're reading or decoding AMF data, you should be prepared to handle either **ECMA Array** or **Object** for greater flexibility and robustness.

- **Default values**: Unless explicitly called out, there should be no assumptions made regarding default values, such as null or undefined.
- **Legacy vs. Enhanced Properties**: In the documentation, an effort has been made to distinguish between legacy properties and newly defined ones through color coding, such as using bold text or different background colors for enhancements. While this color coding is not guaranteed to be consistent, the distinctions between values defined in E-RTMP should be readily apparent.
- **Capability flags:** The capabilities flags, exchanged during a connect handshake, may not cover all possible functionalities. For instance, a client might indicate support for multitrack processing without specifying its ability to encode or decode multitrack streams. In scenarios where a client, capable of issuing a play command, declares multitrack support, it MUST be equipped to handle the playback of such streams. Similarly, if a client is aware of the server's multitrack capabilities, it MAY opt to publish a multitrack stream.

Contextualizing Enhancements

In the following section, we'll outline key enhancements. The aim is to give readers a clear snapshot of the E-RTMP objectives and intentions before diving into the rest of the detailed specifications.

- Newly introduced codecs

Table: Additional codecs for E-RTMP

Additional Audio Codec	Notes
AC-3	AC-3 and E-AC-3 have significantly influenced the surround sound market by offering versatile and scalable audio solutions for both physical and streaming media. Their balance of complexity and performance makes them enduring standards in multichannel audio technology.
E-AC-3	
Opus	Popular in both hardware and software streaming solutions, the [WebCodecs] audio codec registry also includes support for these widely used audio formats.
FLAC	
AAC (added FOURCC signaling)	
MP3 (added FOURCC signaling)	
Additional Video Codec	
AVC (a.k.a., H.264, added FOURCC signaling)	Popular in both hardware and software streaming solutions, the [WebCodecs] video codec registry also includes support for these widely used video formats.
HEVC (a.k.a., H.265)	

VP8 (webRTC officially supports this codec)	
VP9	
AV1	

- HDR - to accommodate new video codecs and cater to the existing spectrum of displays
- VideoPacketType.Metadata - to accommodate diverse video metadata types
- Multichannel configuration - to specify the number of channels and their sequence
- Multitrack - to provide the ability to manage or process multiple audio or video tracks
- and more…

Simple Data Types

The following data types are used in [RTMP] bitstreams and [FLV] files. FOURCC was introduced to support E-RTMP and FLV.

Table: Simple data types

Type	Definition
0x...	Hexadecimal value
UB[n]	Bit field with unsigned n-bit integer, where n is in the range 1 to 31, excluding 8, 16, 24
FOURCC	Four-character ASCII code, such as '**av01**', encoded as UI32
SI8	Signed 8-bit integer
SI16	Signed 16-bit integer
SI24	Signed 24-bit integer
SI32	Signed 32-bit integer
UI8	Unsigned 8-bit integer
UI16	Unsigned 16-bit integer
UI24	Unsigned 24-bit integer
UI32	Unsigned 32-bit integer
xxx[]	Array of type xxx. Number of elements to be inferred
xxx[n]	Array of n elements of type xxx
[xxx]	Array of one element of type xxx

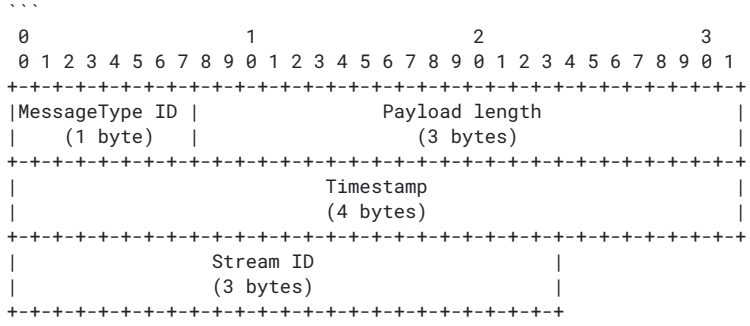
Note: Unless specifically called out, multi-byte integers SHALL be stored in big-endian byte order

RTMP Message Format

Adobe's Real-Time Messaging Protocol (RTMP) is an application-level protocol designed for the multiplexing and packetizing of multimedia streams—such as audio, video, and interactive content, for transmission over network protocols like TCP. A fundamental feature of [RTMP] is the Chunk Stream, which facilitates the multiplexing, packetizing, and prioritization of messages, integral to the protocol's real-time capabilities.

The legacy RTMP specification in Section 6.1 elaborates on the RTMP Message Format, providing precise encoding guidelines for the RTMP message header, inclusive of field widths and byte order. However, this portrayal might be somewhat confusing because RTMP messages, when transported over the Chunk Stream, don't literally conform to this depicted format. An RTMP Message is divided into two principal components: a message virtual header and a message payload. The 'virtual' descriptor indicates that while RTMP messages are carried within the RTMP Chunk Stream, their headers are conceptually encoded as Chunk Message Headers. When these are decoded from the RTMP Chunk Stream, the underlying transport layer, the resulting format is to be understood as a virtual header. This abstract representation aligns with the structured format and semantics detailed in the legacy RTMP specification. Detailed next is the format of the message virtual header and some additional related information.

- Message virtual header



- There are two message types reserved for media messages:
  - The message type value of 8 is reserved for audio message
  - The message type value of 9 is reserved for video messages
- The message payload follows the header and may contain various types of content, such as compressed audio or video data. RTMP itself does not recognize or process the payload's content. If new codec types are to be added, they must be defined where the actual payload internals are outlined. FLV is a container file format where the specifics of the AV payload, including the codecs, are defined.
- Please refer to the legacy RTMP specification (in various locations) and the legacy [FLV] specification (Annex E) for details on the endianness (a.k.a., byte order) of the data format on the wire.

An Overview of the FLV File Format

[FLV] file is a container for AV (Audio and Video) data. The file consists of alternating back-pointers and tags, each accompanied by data related to that tag. Each TagType within a FLV file is unsigned and defined by 5 bits. AUDIODATA has a TagType of 8, and VIDEODATA has a TagType of 9. Note: These TagTypes map to the same values of **MessageType ID**, defined by UI8, in the legacy [RTMP] specification. This alignment is by design.

Tag Types of 8 or 9 are accompanied by an AudioTagHeader or VideoTagHeader. It's common to think of RTMP in conjunction with FLV. However, RTMP is a protocol, and [FLV] is a file container. This distinction is why they are originally defined in separate specifications. This enhancement spec aims to improve both RTMP and FLV.

Pre 2023 AudioTagHeader Format

Below is the AudioTagHeader format for the pre 2023 (a.k.a., legacy) FLV specification:

Table: FLV specification AudioTagHeader

Field	Type	Comment
SoundFormat	UB[4]	Format of SoundData. The following values are defined: 0 = Linear PCM, platform endian 1 = ADPCM 2 = MP3 3 = Linear PCM, little endian 4 = Nellymoser 16 kHz mono 5 = Nellymoser 8 kHz mono 6 = Nellymoser 7 = G.711 A-law logarithmic PCM 8 = G.711 mu-law logarithmic PCM 9 = Reserved 10 = AAC 11 = Speex 12 = Reserved 13 = Reserved 14 = MP3 8 kHz 15 = Device-specific sound Formats 7, 8, 14, and 15 are reserved. AAC is supported in Flash Player 9,0,115,0 and higher. Speex is supported in Flash Player 10 and higher.
SoundRate	UB[2]	Sampling rate. The following values are defined: 0 = 5.5 kHz 1 = 11 kHz 2 = 22 kHz 3 = 44 kHz

SoundSize	UB[1]	Size of each audio sample. This parameter only pertains to uncompressed formats. Compressed formats always decode to 16 bits internally. 0 = 8-bit samples 1 = 16-bit samples
SoundType	UB[1]	Mono or stereo sound 0 = Mono sound 1 = Stereo sound
AACPacketType	IF SoundFormat == 10 UI8	The following values are defined: 0 = AAC sequence header 1 = AAC raw

Pre 2023 VideoTagHeader Format

Below is the VideoTagHeader format for the pre 2023 (a.k.a., legacy) FLV specification:

Table: FLV specification VideoTagHeader

Field	Type	Comment
Frame Type	UB[4]	Type of video frame. The following values are defined: 1 = key frame (for AVC, a seekable frame) 2 = inter frame (for AVC, a non-seekable frame) 3 = disposable inter frame (H.263 only) 4 = generated key frame (reserved for server use only) 5 = video info/command frame
CodecID	UB[4]	Codec Identifier. The following values are defined: 2 = Sorenson H.263 3 = Screen video 4 = On2 VP6 5 = On2 VP6 with alpha channel 6 = Screen video version 2 7 = AVC
AVCPacketType	IF CodecID == 7 UI8	The following values are defined: 0 = AVC sequence header 1 = AVC NALU 2 = AVC end of sequence (lower level NALU sequence ender is not REQUIRED or supported)
CompositionTime	IF CodecID == 7 SI24	IF AVCPacketType == 1 Composition time offset ELSE 0 See ISO/IEC 14496-12, 8.15.3 for an explanation of composition times. The offset in an FLV file is always in milliseconds.

Enhancements to RTMP and FLV

Within the following sections, this document provides a comprehensive overview of the enhancements made to [RTMP] and [FLV]. These improvements are discussed in detail, highlighting their impact and benefits.

Enhancing onMetaData

[FLV] metadata SHALL be encapsulated within a SCRIPTDATA segment, which includes a ScriptTagBody encoded in the Action Message Format (AMF). Importantly, this metadata SHALL always remain unencrypted, even when the FLV content itself is encrypted. This design choice is essential for allowing various FLV parsers to successfully stream the FLV content and for enabling media players to provide contextual information to the user.

The **ScriptTagBody** is structured to encapsulate method invocations. It consists of an item containing a method name (e.g., **onMetaData**) along with a corresponding set of arguments.

To signal FLV metadata, the item within the **ScriptTagBody** MUST encapsulate the method name **onMetaData**, along with a single argument of type ECMA array. This array holds metadata properties, the availability of which may vary depending on the software used to create the FLV. Typical **onMetaData** argument properties include, but are not limited to:

Table: Typical properties found in the **onMetaData** argument object

Property	Type	Comment
audiocodecid	number	Audio codec ID used in the file: See AudioTagHeader of the legacy [FLV] specification for available CodecID values. When [FourCC] is used to signal the codec, this property is set to a FOURCC value. Note: A FOURCC value is big endian relative to the underlying ASCII character sequence (e.g., 'Opus' == 0x4F707573 == 1332770163.0).
audiodatarate	number	Audio bitrate, in kilobits per second
audiodelay	number	Delay introduced by the audio codec, in seconds
audiosamplerate	number	Frequency at which the audio stream is replayed
audiosamplesize	number	Resolution of a single audio sample
canSeekToEnd	boolean	Indicating the last video frame is a key frame
creationdate	string	Creation date and time
duration	number	Total duration of the file, in seconds
filesize	number	Total size of the file, in bytes
framerate	number	Number of frames per second
height	number	Height of the video, in pixels
stereo	boolean	Indicates stereo audio

videocodecid	number	Video codec ID used in the file: See VideoTagHeader of the legacy [FLV] specification for available CodecID values. When [FourCC] is used to signal the codec, this property is set to a FOURCC value. Note: A FOURCC value is big endian relative to the underlying ASCII character sequence (e.g., 'av01' == 0x61763031 == 1635135537.0).
videodatarate	number	Video bitrate, in kilobits per second
width	number	Width of the video, in pixels

Note: The properties **audiocodecid** and **videocodecid** have been enhanced to support FOURCC (Four-byte ASCII code) values. These values are interpreted as UI32 (e.g., 'av01').

Reconnect Request

Objective

[RTMP] packetizes multimedia streams using a suitable transport protocol, typically a persistent TCP connection. There are instances when a streaming platform may request the streaming client to reconnect, such as:

- When live streaming servers undergo updates.
- When there's a need to redirect the client to a different server instance, ensuring optimal load balancing and precise geolocation mapping.

To accommodate these needs, a **NetConnection.Connect.ReconnectRequest** status event has been introduced as part of the **NetConnection onStatus** command.

NetConnection Commands

NetConnection establishes a bidirectional link between a client and a server, allowing for asynchronous Remote Procedure Calls (RPCs). The following commands (a.k.a., predefined RPCs) can be issued via **NetConnection**:

- connect
- createStream
- deleteStream
- onStatus

The **onStatus** command has been enhanced to include the capability to request a client to reconnect. Servers can issue an **onStatus** command to prompt clients to adapt to changes in **NetConnection** status. The structure of this command, as relayed from the server to the client, is outlined below:

Table: Server to client, NetConnection onStatus command

Field Name	Type	Description
Command Name	string	Name of the command. Set to "onStatus"
Transaction ID	number	Transaction ID set to 0. (i.e., no response needed)
Command Object	null	There is no command object for **onStatus** command.
Info Object	Object	An AMF-encoded object, the properties of which are utilized by the **onStatus** command. The Info Object provides information about the status of the current connection.

The following is a description of AMF-encoded name-value pairs in the Info Object for the **onStatus** command when handling reconnect. It MAY contain other properties as appropriate to the client.

Table: Info Object parameter for onStatus command when handling reconnect

Property	Type	Description	Example Value
tcUrl (optional)	string	Absolute or relative URI reference of the server to which to reconnect. If not specified, use the tcUrl for the current connection. A relative URI reference should be resolved relative to the tcUrl for the current connection.	1. rtmp://foo.mydomain.com:1935/realtimeapp 2. rtmp://127.0.0.1/realtimeapp 3. //192.0.2.0/realtimeapp 4. /realtimeapp
code	string	A string identifying the event that occurred. To reconnect **code** MUST be set to "NetConnection.Connect.ReconnectRequest"	NetConnection.Connect.ReconnectRequest
description (optional)	string	A string containing human-readable information about the message. Not every information object includes this property.	The streaming server is undergoing updates.
level	string	A string indicating the severity of the event. To reconnect the **level** MUST be set to "status".	status

Message Flow When Handling NetConnection.Connect.ReconnectRequest

1. Prior to the shutdown of the live streaming server or when the server intends to remap the client to another server instance, it dispatches an **onStatus** command to the client with a **code** of **NetConnection.Connect.ReconnectRequest**. If the server aims to remap the client, it MUST set the **tcUrl** property in the Info Object. In order to avoid a disruption, the server managing the original connection (commonly referred to as the "old server") SHOULD continue processing messages from the client until the client disconnects.
2. When the client receives the **NetConnection.Connect.ReconnectRequest** event, it persists in streaming to/from the current server up to the next appropriate media boundary, such as a keyframe. Subsequently, it establishes a connection with a new server and disconnects from the old server. If the Info Object includes the **tcUrl** property, the client uses this URL for the reconnection process. Absent this property, the client defaults to the **tcUrl** for the current connection.
3. While the client can establish a new connection before severing the original one, it SHOULD exercise caution to ensure the Quality of Service (QoS) is not compromised.

The capability to support the **NetConnection.Connect.ReconnectRequest** event becomes evident during the initial connect phase. Detailed guidelines for signaling reconnect ability can be found in the Enhancing NetConnection **connect** Command section.

Detailed Overview of the onStatus Command for NetConnection

The server-to-client **onStatus** command for **NetConnection**, serves a crucial function within the RTMP framework. Though the legacy RTMP specification may not have detailed this command, the goal here is to offer an overview for a better understanding.

Both clients and servers can initiate RPCs at the receiving end, with some RPCs being predefined as commands. **onStatus** stands out as one such essential command.

When using the **onStatus** command, the goal is to inform the client about the status of the connection. Each dispatched command message comprises the following elements:

- Command Name: type **string**
- Transaction ID: type **number**
- Command Object (set to null when dispatching an onStatus command): type **Object**
- Info Object (which can be viewed as Optional Arguments): type **Object**

Both the Command Object and the Info Object offer additional context and details for the command. The **onStatus** command is triggered whenever there's a status change or an error concerning the **NetConnection**. To handle this information, you should define a callback function.

```js
// Sample pseudocode for the onStatus callback function
nc.onStatus = function(infoObject) {
  // Handle the status change or error here.
}
```

infoObject is an AMF-encoded object with properties that provide information about the status of a **NetConnection**. It contains at least the following three properties, but MAY contain other properties as appropriate to the client.

Table: **infoObject** for **onStatus** command

Property	Type	Description	Example Value
code	string	A string identifying the event that occurred.	NetConnection.Connect.Success
description (optional)	string	A string containing human-readable information about the message. Not every information object includes this property.	The connection attempt succeeded.
level	string	There are three established values for level: "status", "warning", and "error".	status

The table below provides examples of **code**, **level**, and **description** property values. Please note that this is not an exhaustive list, and not all entries may apply to every type of client. Additionally, the **description** property values included are merely illustrative examples; developers are responsible for conveying the appropriate meaning in their specific solutions.

Table: **code**, **level** and **description** values for **infoObject** used by **onStatus**

Code	Level	Description

NetConnection.Call.Failed	error	The NetConnection.call() method was not able to invoke the server-side method or command.
NetConnection.Connect.AppShutdown	error	The application has been shut down (for example, if the application is out of memory resources and must shut down to prevent the server from crashing) or the server has shut down.
NetConnection.Connect.Closed	status	The connection was closed successfully.
NetConnection.Connect.Failed	error	The connection attempt failed.
NetConnection.Connect.Rejected	error	The client does not have permission to connect to the application.
NetConnection.Connect.Success	status	The connection attempt succeeded.
NetConnection.Connect.ReconnectRequest	**status**	**The server is requesting the client to reconnect.**
NetConnection.Proxy.NotResponding	error	The proxy server is not responding. See the ProxyStream class.

Enhanced Audio

The AudioTagHeader has been extended to define additional audio codecs, multichannel audio, multitrack capabilities, signaling support, and additional miscellaneous enhancements, while ensuring backward compatibility. This extension is termed the ExAudioTagHeader and is designed to be future-proof, allowing for the definition of additional audio codecs, features, and corresponding signaling.

During the parsing process, the logic MUST handle unexpected or unknown elements gracefully. Specifically, if any critical signaling or flags (e.g., AudioPacketType and AudioFourCc) are not recognized, the system MUST fail in a controlled and predictable manner.

IMPORTANT: A single audio message for a unique timestamp may include a batch of AudioPacketType values (e.g., multiple TrackId values). When parsing an audio message, the bitstream MUST be processed completely to ensure all payload data has been handled.

Table: Extended AudioTagHeader

Description Of Bitstream	Enumerated Types

soundFormat = UB[4] as SoundFormat
 	
if (soundFormat != SoundFormat.ExHeader) {
  // See AudioTagHeader of the legacy [FLV] specification for for detailed format
  // of the four bits used for soundRate/soundSize/soundType
  //
  // Note: soundRate, soundSize and soundType formats have not changed.
  // if (soundFormat == SoundFormat.ExHeader) we switch into FOURCC audio mode
  // as defined below. This means that soundRate, soundSize and soundType
  // bits are not interpreted, instead the UB[4] bits are interpreted as an
  // AudioPacketType
  soundRate = UB[2]
  soundSize = UB[1]
  soundType = UB[1]
}
	enum SoundFormat {
	  LPcmPlatformEndian = 0,
	  AdPcm = 1,
	  Mp3 = 2,
	  LPcmLittleEndian = 3,
	  Nellymoser16KMono = 4,
	  Nellymoser8KMono = 5,
	  Nellymoser = 6,
	  G711ALaw = 7,
	  G711MuLaw = 8,
	  ExHeader = 9, // new, used to signal FOURCC mode
	  Aac = 10,
	  Speex = 11,
	  // 12 - reserved
	  // 13 - reserved
	  Mp3_8K = 14,
	  Native = 15, // Device specific sound
	}

ExAudioTagHeader Section	

Note: **ExAudioTagHeader** is present if (soundFormat == SoundFormat.**ExHeader**)

Description Of Bitstream

```
//
// process ExAudioTagHeader
//
processAudioBody = false
if (soundFormat == SoundFormat.ExHeader) {
  processAudioBody = true

  // The UB[4] bits are interpreted as AudioPacketType
  // instead of sound rate, size and type
  audioPacketType = UB[4] as AudioPacketType

  if (audioPacketType == AudioPacketType.Multitrack) {
    isAudioMultitrack = true;
    audioMultitrackType = UB[4] as AvMultitrackType

    // Fetch AudioPacketType for all audio tracks in the audio message.
    // This fetch MUST not result in a AudioPacketType.Multitrack
    audioPacketType = UB[4] as AudioPacketType

    if (audioMultitrackType != AvMultitrackType.ManyTracksManyCodecs) {
      // The tracks are encoded with the same codec. Fetch the FOURCC for them
      audioFourCc = FOURCC as AudioFourCc
    }
  } else {
    audioFourCc = FOURCC as AudioFourCc
  }
}
```

Enumerated Types

```
enum AudioPacketType {
  SequenceStart       = 0,
  CodedFrames         = 1,

  // RTMP includes a previously undocumented 'audio silence' message.
  // This silence message is identified when an audio message contains
  // a zero-length payload, or more precisely, an empty audio message
  // without an AudioTagHeader, indicating a period of silence. The
  // action to take after receiving a silence message is system
  // dependent. The semantics of the silence message in the Flash
  // Media playback and timing model are as follows:
  //
  // - Ensure all buffered audio data is played out before entering the
  //   silence period:
  //   Make sure that any audio data currently in the buffer is fully
  //   processed and played. This ensures a clean transition into the
  //   silence period without cutting off any audio.
  //
  // - After playing all buffered audio data, flush the audio decoder:
  //   Clear the audio decoder to reset its state and prepare it for new
  //   input after the silence period.
  //
  // - During the silence period, the audio clock can't be used as the
  //   master clock for synchronizing playback:
  //   Switch to using the system's wall-clock time to maintain the correct
  //   timing for video and other data streams.
  //
  // - Don't wait for audio frames for synchronized A+V playback:
  //   Normally, audio frames drive the synchronization of audio and video
  //   (A/V) playback. During the silence period, playback should not stall
  //   waiting for audio frames. Video and other data streams should
  //   continue to play based on the wall-clock time, ensuring smooth
  //   playback without audio.
  //
  // AudioPacketType.SequenceEnd is to have no less than the same meaning as
  // a silence message. While it may seem redundant, we need to introduce
  // this enum to ensure we can signal the end of the audio sequence for any
  // audio track.
  SequenceEnd         = 2,

  //  3 - Reserved

  MultichannelConfig  = 4,

  // Turns on audio multitrack mode
  Multitrack          = 5,

  //  6 - Reserved
  // ...
```

	// 14 - reserved
	// 15 - reserved
	}
	
	enum AudioFourCc {
	//
	// Valid FOURCC values for signaling support of audio codecs
	// in the enhanced FourCC pipeline. In this context, support
	// for a FourCC codec MUST be signaled via the enhanced
	// 'connect' command.
	//
	
	// AC-3/E-AC-3 - <https://en.wikipedia.org/wiki/Dolby_Digital>
	Ac3 = makeFourCc('ac-3'),
	Eac3 = makeFourCc('ec-3'),
	
	// Opus audio - <https://opus-codec.org/>
	Opus = makeFourCc('Opus'),
	
	// Mp3 audio - <https://en.wikipedia.org/wiki/MP3>
	Mp3 = makeFourCc('.mp3'),
	
	// Free Lossless Audio Codec - <https://xiph.org/flac/format.html>
	Flac = makeFourCc('fLaC'),
	
	// Advanced Audio Coding - <https://en.wikipedia.org/wiki/Advanced_Audio_Coding>
	// The following AAC profiles, denoted by their object types, are supported
	// 1 = main profile
	// 2 = low complexity, a.k.a., LC
	// 5 = high efficiency / scale band replication, a.k.a., HE / SBR
	Aac = makeFourCc('mp4a'),
	}
	
	enum AvMultitrackType {
	//
	// Used by audio and video pipeline
	//
	
	OneTrack = 0,
	ManyTracks = 1,
	ManyTracksManyCodecs = 2,
	
	// 3 - Reserved
	// ...
	// 15 - Reserved
	}

ExAudioTagBody Section Note: This ExAudioTagBody format is signaled by the presence of **ExAudioTagHeader**	
Description Of Bitstream	Enumerated Types
// // process ExAudioTagBody	enum AudioChannelOrder { //

```
//
while (processAudioBody) {
  if (isAudioMultitrack) {
    if (audioMultitrackType == AvMultitrackType.ManyTracksManyCodecs) {
      // Each track has a codec assigned to it. Fetch the FOURCC for the next track.
      audioFourCc = FOURCC as AudioFourCc
    }

    // Track Ordering:
    //
    // For identifying the highest priority (a.k.a., default track)
    // or highest quality track, it is RECOMMENDED to use trackId
    // set to zero. For tracks of lesser priority or quality, use
    // multiple instances of trackId with ascending numerical values.
    // The concept of priority or quality can have multiple
    // interpretations, including but not limited to bitrate,
    // resolution, default angle, and language. This recommendation
    // serves as a guideline intended to standardize track numbering
    // across various applications.
    audioTrackId = UI8

    if (audioMultitrackType != AvMultitrackType.OneTrack) {
      // The 'sizeOfAudioTrack' specifies the size in bytes of the
      // current track that is being processed. This size starts
      // counting immediately after the position where the 'sizeOfAudioTrack'
      // value is located. You can use this value as an offset to locate the
      // next audio track in a multitrack system. The data pointer is
      // positioned immediately after this field. Depending on the MultiTrack
      // type, the offset points to either a 'fourCc' or a 'trackId.'
      sizeOfAudioTrack = UI24
    }
  }

  if (audioPacketType == AudioPacketType.MultichannelConfig) {
    //
    // Specify a speaker for a channel as it appears in the bitstream.
    // This is needed if the codec is not self-describing for channel mapping
    //

    // set audio channel order
    audioChannelOrder = UI8 as AudioChannelOrder

    // number of channels
    channelCount = UI8

    if (audioChannelOrder == AudioChannelOrder.Custom) {
      // Each entry specifies the speaker layout (see AudioChannel enum above
      // for layout definition) in the order that it appears in the bitstream.
      // First entry (i.e., index 0) specifies the speaker layout for channel 1.
      // Subsequent entries specify the speaker layout for the next channels
      // (e.g., second entry for channel 2, third entry for channel 3, etc.).
      audioChannelMapping = UI8[channelCount] as AudioChannel
```

```
  // Only the channel count is specified, without any further information
  // about the channel order
  //
  Unspecified = 0,

  //
  // The native channel order (i.e., the channels are in the same order in
  // which as defined in the AudioChannel enum).
  //
  Native      = 1,

  //
  // The channel order does not correspond to any predefined
  // order and is stored as an explicit map.
  //
  Custom      = 2
}

num AudioChannelMask {
  //
  // Mask used to indicate which channels are present in the stream.
  //

  // masks for commonly used speaker configurations
  // <https://en.wikipedia.org/wiki/Surround_sound#Standard_speaker_channels>
  FrontLeft           = 0x000001,
  FrontRight          = 0x000002,
  FrontCenter         = 0x000004,
  LowFrequency1       = 0x000008,
  BackLeft            = 0x000010,
  BackRight           = 0x000020,
  FrontLeftCenter     = 0x000040,
  FrontRightCenter    = 0x000080,
  BackCenter          = 0x000100,
  SideLeft            = 0x000200,
  SideRight           = 0x000400,
  TopCenter           = 0x000800,
  TopFrontLeft        = 0x001000,
  TopFrontCenter      = 0x002000,
  TopFrontRight       = 0x004000,
  TopBackLeft         = 0x008000,
  TopBackCenter       = 0x010000,
  TopBackRight        = 0x020000,

  // Completes 22.2 multichannel audio,
  // as standardized in SMPTE ST2036-2-2008
  // see - <https://en.wikipedia.org/wiki/22.2_surround_sound>
  LowFrequency2       = 0x040000,
  TopSideLeft         = 0x080000,
  TopSideRight        = 0x100000,
  BottomFrontCenter   = 0x200000,
  BottomFrontLeft     = 0x400000,
```

```
  }

  if (audioChannelOrder == AudioChannelOrder.Native) {
    // audioChannelFlags indicates which channels are present in the
    // multi-channel stream. You can perform a Bitwise AND
    // (i.e., audioChannelFlags & AudioChannelMask.xxx) to see if a
    // specific audio channel is present
    audioChannelFlags = UI32
  }
}

if (audioPacketType == AudioPacketType.SequenceEnd) {
  // signals end of sequence
}

if (audioPacketType == AudioPacketType.SequenceStart) {
  if (audioFourCc == AudioFourCc.Aac) {
    // The AAC audio specific config (a.k.a., AacSequenceHeader) is
    // defined in ISO/IEC 14496-3.
    aacHeader = [AacSequenceHeader]
  }

  if (audioFourCc == AudioFourCc.Flac) {
    // FlacSequenceHeader layout is:
    //
    // The bytes 0x66 0x4C 0x61 0x43 ("fLaC" in ASCII) signature
    //
    // Followed by a metadata block (called the STREAMINFO block) as described
    // in section 7 of the FLAC specification. The STREAMINFO block contains
    // information about the whole sequence, such as sample rate, number of
    // channels, total number of samples, etc. It MUST be present as the first
    // metadata block in the sequence. The FLAC audio specific bitstream format
    // is defined at <https://xiph.org/flac/format.html>
    flacHeader = [FlacSequenceHeader]
  }

  if (audioFourCc == AudioFourCc.Opus) {
    // Opus Sequence header (a.k.a., ID header):
    // - The Opus sequence start is also known as the ID header.
    // - It contains essential information needed to initialize
    //   the decoder and understand the stream format.
    // - For detailed structure, refer to RFC 7845, Section 5.1:
    //   <https://datatracker.ietf.org/doc/html/rfc7845#section-5.1>
    //
    // If the Opus sequence start payload is empty, use the
    // AudioPacketType.MultichannelConfig signal for channel
    // mapping when present; otherwise, default to mono/stereo mode.
    opusHeader = [OpusSequenceHeader]
  }
}

if (audioPacketType == AudioPacketType.CodedFrames) {
```

```
  BottomFrontRight    = 0x800000,
}

enum AudioChannel {
  //
  // Channel mappings enums
  //

  // commonly used speaker configurations
  // see - <https://en.wikipedia.org/wiki/Surround_sound#Standard_speaker_channels>
  FrontLeft           = 0,  // i.e., FrontLeft is assigned to channel zero
  FrontRight,
  FrontCenter,
  LowFrequency1,
  BackLeft,
  BackRight,
  FrontLeftCenter,
  FrontRightCenter,
  BackCenter          = 8,
  SideLeft,
  SideRight,
  TopCenter,
  TopFrontLeft,
  TopFrontCenter,
  TopFrontRight,
  TopBackLeft,
  TopBackCenter       = 16,
  TopBackRight,

  // mappings to complete 22.2 multichannel audio, as
  // standardized in SMPTE ST2036-2-2008
  // see - <https://en.wikipedia.org/wiki/22.2_surround_sound>
  LowFrequency2       = 18,
  TopSideLeft,
  TopSideRight,
  BottomFrontCenter,
  BottomFrontLeft,
  BottomFrontRight,

  // Channel is empty and can be safely skipped.
  Unused              = 0xfe,

  // Channel contains data, but its speaker configuration is unknown.
  Unknown             = 0xff,
}
```

```
  if (audioFourCc == AudioFourCc.Ac3 || audioFourCc == AudioFourCc.Eac3) {
    // Body contains audio data as defined by the bitstream syntax
    // in the ATSC standard for Digital Audio Compression (AC-3, E-AC-3)
    ac3Data = [Ac3CodedData]
  }

  if (audioFourCc == AudioFourCc.Opus) {
    // Body contains Opus packets. The layout is one Opus
    // packet for each of N different streams, where N is
    // typically one for mono or stereo, but MAY be greater
    // than one for multichannel audio. The value N is
    // specified in the ID header (Opus sequence start) or
    // via the AudioPacketType.MultichannelConfig signal, and
    // is fixed over the entire length of the Opus sequence.
    // The first (N - 1) Opus packets, if any, are packed one
    // after another using the self-delimiting framing from
    // Appendix B of [RFC6716]. The remaining Opus packet is
    // packed at the end of the Ogg packet using the regular,
    // undelimited framing from Section 3 of [RFC6716]. All
    // of the Opus packets in a single audio packet MUST be
    // constrained to have the same duration.
    opusData = [OpusCodedData]
  }

  if (audioFourCc == AudioFourCc.Mp3) {
    // An Mp3 audio stream is built up from a succession of smaller
    // parts called frames. Each frame is a data block with its own header
    // and audio information
    mp3Data = [Mp3CodedData]
  }

  if (audioFourCc == AudioFourCc.Aac) {
    // The AAC audio specific bitstream format is defined in ISO/IEC 14496-3.
    aacData = [AacCodedData]
  }

  if (audioFourCc == AudioFourCc.Flac) {
    // The audio data is composed of one or more audio frames. Each frame
    // consists of a frame header, which contains a sync code and information
    // about the frame, such as the block size, sample rate, number of
    // channels, et cetera. The Flac audio specific bitstream format
    // is defined at <https://xiph.org/flac/format.html>
    flacData = [FlacCodedData]
  }
}

if (isAudioMultitrack &&
    audioMultitrackType != AvMultitrackType.OneTrack &&
    positionDataPtrToNextAudioTrack(sizeOfAudioTrack)) {
    // positionDataPtrToNextAudioTrack() is for developer to write
    continue
}
```

// done processing audio message
break
}	

Enhanced Video

The VideoTagHeader has been extended to define additional video codecs, multitrack capabilities, signaling support, and additional miscellaneous enhancements, while ensuring backward compatibility. This extension is termed the ExVideoTagHeader and is designed to be future-proof, allowing for the definition of additional video codecs, features, and corresponding signaling.

During the parsing process, the logic MUST handle unexpected or unknown elements gracefully. Specifically, if any critical signaling or flags (e.g., VideoFrameType, VideoPacketType, or VideoFourCc) are not recognized, the system MUST fail in a controlled and predictable manner.

IMPORTANT: A single video message for a unique timestamp may include a batch of VideoPacketType values (e.g., multiple TrackId values, Metadata values). When parsing a video message, the bitstream MUST be processed completely to ensure all payload data has been handled.

Table: Extended VideoTagHeader

Description Of Bitstream	Enumerated Types

// Check if isExVideoHeader flag is set to 1, signaling E-RTMP
// video mode. In this case, VideoCodecId's 4-bit unsigned binary (UB[4])
// should not be interpreted as a codec identifier. Instead, these
// UB[4] bits should be interpreted as VideoPacketType.
isExVideoHeader = UB[1]
videoFrameType = UB[3] as VideoFrameType
 	
if (isExVideoHeader == 0) {
  // Utilize the VideoCodecId values and the bitstream description
  // as defined in the legacy [FLV] specification. Refer to this
  // version for the proper implementation details.
  videoCodecId = UB[4] as VideoCodecId
 	
  if (videoFrameType == VideoFrameType.Command) {
  videoCommand = UI8 as VideoCommand
  }
}
	enum VideoFrameType {
	  // 0 - reserved
	  KeyFrame = 1, // a seekable frame
	  InterFrame = 2, // a non - seekable frame
	  DisposableInterFrame = 3, // H.263 only
	  GeneratedKeyFrame = 4, // reserved for server use only
	
	  // If videoFrameType is not ignored and is set to VideoFrameType.Command,
	  // the payload will not contain video data. Instead, (Ex)VideoTagHeader
	  // will be followed by a UI8, representing the following meanings:
	  //
	  // 0 = Start of client-side seeking video frame sequence
	  // 1 = End of client-side seeking video frame sequence
	  //
	  // frameType is ignored if videoPacketType is VideoPacketType.MetaData
	  Command = 5, // video info / command frame
	
	  // 6 = reserved
	  // 7 = reserved
	}
	
	enum VideoCommand {
	  StartSeek = 0,
	  EndSeek = 1,

```
  // 0x03 = reserved
  // ...
  // 0xff = reserved
}

enum VideoCodecId {
    // These values remain as they were in the legacy [FLV] specification.
    // If the IsExVideoHeader flag is set, we switch into
    // FOURCC video mode defined in the VideoFourCc enumeration.
    // This means that VideoCodecId (UB[4] bits) is not interpreted
    // as a codec identifier. Instead, these UB[4] bits are
    // interpreted as VideoPacketType.

    // 0 - Reserved
    // 1 - Reserved
    SorensonH263    = 2,
    Screen          = 3,
    On2VP6          = 4,
    On2VP6A         = 5, // with alpha channel
    ScreenV2        = 6,
    Avc             = 7,
    // 8 - Reserved
    // ...
    // 15 - Reserved
}
```

ExVideoTagHeader Section

note: **ExVideoTagHeader** is present if **IsExVideoHeader** flag is set.

Description Of Bitstream

```
//
// process ExVideoTagHeader
//
processVideoBody = false
if (isExVideoHeader == 1) {
  processVideoBody = true

  // The UB[4] bits are interpreted as VideoPacketType
  // instead of VideoCodecId
  videoPacketType = UB[4] as VideoPacketType

  if (videoPacketType != VideoPacketType.Metadata &&
    videoFrameType == VideoFrameType.Command) {
    videoCommand = UI8 as VideoCommand

    // ExVideoTagBody has no payload if we got here.
    // Set boolean to not try to process the video body.
    processVideoBody = false
  } else if (videoPacketType == VideoPacketType.Multitrack) {
    isVideoMultitrack = true;
    videoMultitrackType = UB[4] as AvMultitrackType

    // Fetch VideoPacketType for all video tracks in the video message.
```

Enumerated Types

```
enum VideoPacketType {
  SequenceStart         = 0,
  CodedFrames           = 1,
  SequenceEnd           = 2,

  // CompositionTime Offset is implicitly set to zero. This optimization
  // avoids transmitting an SI24 composition time value of zero over the wire.
  // See the ExVideoTagBody section below for corresponding pseudocode.
  CodedFramesX          = 3,

  // ExVideoTagBody does not contain video data. Instead, it contains
  // an AMF-encoded metadata. Refer to the Metadata Frame section for
  // an illustration of its usage. For example, the metadata might include
  // HDR information. This also enables future possibilities for expressing
  // additional metadata meant for subsequent video sequences.
  //
  // If VideoPacketType.Metadata is present, the FrameType flags
  // at the top of this table should be ignored.
  Metadata              = 4,

  // Carriage of bitstream in MPEG-2 TS format
  //
  // PacketTypeSequenceStart and PacketTypeMPEG2TSSequenceStart
```

```
    // This fetch MUST not result in a VideoPacketType.Multitrack
    videoPacketType = UB[4] as VideoPacketType

    if (videoMultitrackType != AvMultitrackType.ManyTracksManyCodecs) {
      // The tracks are encoded with the same codec. Fetch the FOURCC for them
      videoFourCc = FOURCC as VideoFourCc
    }
  } else {
    videoFourCc = FOURCC as VideoFourCc
  }
}
```

```
  // are mutually exclusive
  MPEG2TSSequenceStart  = 5,

  // Turns on video multitrack mode
  Multitrack            = 6,

  //  7 - Reserved
  // ...
  // 14 - reserved
  // 15 - reserved
}

enum VideoFourCc {
  //
  // Valid FOURCC values for signaling support of video codecs
  // in the enhanced FourCC pipeline. In this context, support
  // for a FourCC codec MUST be signaled via the enhanced
  // 'connect' command.
  //

  Vp8         = makeFourCc('vp08'),
  Vp9         = makeFourCc('vp09'),
  Av1         = makeFourCc('av01'),
  Avc         = makeFourCc('avc1'),
  Hevc        = makeFourCc('hvc1'),
}

enum AvMultitrackType {
  //
  // Used by audio and video pipeline
  //

  OneTrack              = 0,
  ManyTracks            = 1,
  ManyTracksManyCodecs  = 2,

  //  3 - Reserved
  // ...
  // 15 - Reserved
}
```

ExVideoTagBody Section

Note: This **ExVideoTagBody** format is signaled by the presence of **ExVideoTagHeader** and if **videoCommand** has not been set (see VideoFrameType description)

Description Of Bitstream

```
//
// process ExVideoTagBody
//
while (processVideoBody) {
  if (isVideoMultitrack) {
    if (videoMultitrackType == AvMultitrackType.ManyTracksManyCodecs) {
      // Each track has a codec assigned to it. Fetch the FOURCC for the next track.
      videoFourCc = FOURCC as VideoFourCc
```

```
  }

  // Track Ordering:
  //
  // For identifying the highest priority (a.k.a., default track)
  // or highest quality track, it is RECOMMENDED to use trackId
  // set to zero. For tracks of lesser priority or quality, use
  // multiple instances of trackId with ascending numerical values.
  // The concept of priority or quality can have multiple
  // interpretations, including but not limited to bitrate,
  // resolution, default angle, and language. This recommendation
  // serves as a guideline intended to standardize track numbering
  // across various applications.
  videoTrackId = UI8

  if (videoMultitrackType != AvMultitrackType.OneTrack) {
    // The 'sizeOfVideoTrack' specifies the size in bytes of the
    // current track that is being processed. This size starts
    // counting immediately after the position where the 'sizeOfVideoTrack'
    // value is located. You can use this value as an offset to locate the
    // next video track in a multitrack system. The data pointer is
    // positioned immediately after this field. Depending on the MultiTrack
    // type, the offset points to either a 'fourCc' or a 'trackId.'
    sizeOfVideoTrack = UI24
  }
}

if (videoPacketType == VideoPacketType.Metadata) {
  // The body does not contain video data; instead, it consists of AMF-encoded
  // metadata. The metadata is represented by a series of [name, value] pairs.
  // Currently, the only defined [name, value] pair is ["colorInfo", Object].
  // See the Metadata Frame section for more details on this object.
  //
  // For a deeper understanding of the encoding, please refer to the descriptions
  // of SCRIPTDATA and SCRIPTDATAVALUE in the [FLV] file specification.
  videoMetadata = [VideoMetadata]
}

if (videoPacketType == VideoPacketType.SequenceEnd) {
  // signals end of sequence
}

if (videoPacketType == VideoPacketType.SequenceStart) {
  if (videoFourCc == VideoFourCc.Vp8) {
    // body contains a VP8 configuration record to start the sequence
    vp8Header = [VPCodecConfigurationRecord]
  }

  if (videoFourCc == VideoFourCc.Vp9) {
    // body contains a VP9 configuration record to start the sequence
    vp9Header = [VPCodecConfigurationRecord]
  }
```

```
  if (videoFourCc == VideoFourCc.Av1) {
    // body contains a configuration record to start the sequence
    av1Header = [AV1CodecConfigurationRecord]
  }

  if (videoFourCc == VideoFourCc.Avc) {
    // body contains a configuration record to start the sequence. See ISO/IEC
    // 14496-15, 5.2.4.1 for the description of AVCDecoderConfigurationRecord
    avcHeader = [AVCDecoderConfigurationRecord]
  }

  if (videoFourCc == VideoFourCc.Hevc) {
    // body contains a configuration record to start the sequence. See ISO/IEC
    // 14496-15, 8.3.3.1.2 for the description of HEVCDecoderConfigurationRecord
    hevcHeader = [HEVCDecoderConfigurationRecord]
  }
}

if (videoPacketType == VideoPacketType.MPEG2TSSequenceStart) {
  if (videoFourCc == VideoFourCc.Av1) {
    // body contains a video descriptor to start the sequence
    av1Header = [AV1VideoDescriptor]
  }
}

if (videoPacketType == VideoPacketType.CodedFrames) {
  if (videoFourCc == VideoFourCc.Vp8) {
    // body contains series of coded full frames
    vp8CodedData = [Vp8CodedData]
  }

  if (videoFourCc == VideoFourCc.Vp9) {
    // body contains series of coded full frames
    vp9CodedData = [Vp9CodedData]
  }

  if (videoFourCc == VideoFourCc.Av1) {
    // body contains one or more OBUs which represent a single temporal unit
    av1CodedData = [Av1CodedData]
  }

  if (videoFourCc == VideoFourCc.Avc) {
    // See ISO/IEC 14496-12, 8.15.3 for an explanation of composition times.
    // The offset in an FLV file is always in milliseconds.
    compositionTimeOffset = SI24

    // Body contains one or more NALUs; full frames are required
    avcCodedData = [AvcCodedData]
  }

  if (videoFourCc == VideoFourCc.Hevc) {
```

```
      // See ISO/IEC 14496-12, 8.15.3 for an explanation of composition times.
      // The offset in an FLV file is always in milliseconds.
      compositionTimeOffset = SI24

      // Body contains one or more NALUs; full frames are required
      hevcData = [HevcCodedData]
    }
  }

  if (VideoPacketType.CodedFramesX) {
    // compositionTimeOffset is implied to equal zero. This is
    // an optimization to save putting SI24 value on the wire

    if (videoFourCc == VideoFourCc.Avc) {
      // Body contains one or more NALUs; full frames are required
      avcCodedData = [AvcCodedData]
    }

    if (videoFourCc == VideoFourCc.Hevc) {
      // Body contains one or more NALUs; full frames are required
      hevcData = [HevcCodedData]
    }
  }

  if (isVideoMultitrack &&
      videoMultitrackType != AvMultitrackType.OneTrack &&
      positionDataPtrToNextVideoTrack(sizeOfVideoTrack)) {
      // positionDataPtrToNextVideoTrack() is for developer to write
      continue
  }

  // done processing video message
  break
}
```

Metadata Frame

To support various types of video metadata, the legacy [FLV] specification has been enhanced. The VideoTagHeader has been extended to define a new **VideoPacketType.Metadata** (see ExVideoTagHeader table in Enhanced Video section) whose payload will contain an AMF-encoded metadata. The metadata will be represented by a series of [name, value] pairs. For now the only defined [name, value] pair is ["colorInfo", Object]. When leveraging PacketTypeMetadata to deliver HDR metadata, the metadata MUST be sent prior to the video sequence, scene, frame or such that it affects. Each time a new **colorInfo** object is received it invalidates and replaces the current one. To reset to the original color state you can send **colorInfo** with a value of Undefined (the RECOMMENDED approach) or an empty object (i.e., **{}**).

It is intentional to leverage a video message to deliver PacketTypeMetadata instead of other [RTMP] Message types. One benefit of leveraging a video message is to avoid any racing conditions between video messages and other RTMP message types. Given this, once your **colorInfo**

object is parsed, the read values MUST be processed in time to affect the first frame of the video section which follows the **colorInfo** object.

The **colorInfo** object provides HDR metadata to enable a higher quality image source conforming to BT.2020 (a.k.a., Rec. 2020) standard. The properties of the **colorInfo** object, which are encoded in an AMF message format, are defined below.

Note:

- For content creators: Whenever it behooves to add video hint information via metadata (ex. HDR) to the FLV container it is RECOMMENDED to add it via VideoPacketType.Metadata. This may be done in addition (or instead) to encoding the metadata directly into the codec bitstream.
- The object encoding format (i.e., AMF0 or AMF3) is signaled during the connect command.

```js
type ColorInfo = {
  colorConfig: {
    // number of bits used to record the color channels for each pixel
    bitDepth:                number, // SHOULD be 8, 10 or 12

    //
    // colorPrimaries, transferCharacteristics and matrixCoefficients are defined
    // in ISO/IEC 23091-4/ITU-T H.273. The values are an index into
    // respective tables which are described in "Colour primaries",
    // "Transfer characteristics" and "Matrix coefficients" sections.
    // It is RECOMMENDED to provide these values.
    //

    // indicates the chromaticity coordinates of the source color primaries
    colorPrimaries:          number, // enumeration [0-255]

    // opto-electronic transfer characteristic function (ex. PQ, HLG)
    transferCharacteristics:  number, // enumeration [0-255]

    // matrix coefficients used in deriving luma and chroma signals
    matrixCoefficients:      number, // enumeration [0-255]
  },

  hdrCll: {
    //
    // maximum value of the frame average light level
    // (in 1 cd/m2) of the entire playback sequence
    //
    maxFall:  number,     // [0.0001-10000]

    //
    // maximum light level of any single pixel (in 1 cd/m2)
    // of the entire playback sequence
    //
    maxCLL:   number,     // [0.0001-10000]
  },
```

```
  //
  // The hdrMdcv object defines mastering display (i.e., where
  // creative work is done during the mastering process) color volume (a.k.a., mdcv)
  // metadata which describes primaries, white point and min/max luminance. The
  // hdrMdcv object SHOULD be provided.
  //
  // Specification of the metadata along with its ranges adhere to the
  // ST 2086:2018 - SMPTE Standard (except for minLuminance see
  // comments below)
  //
  hdrMdcv: {
    //
    // Mastering display color volume (mdcv) xy Chromaticity Coordinates within CIE
    // 1931 color space.
    //
    // Values SHALL be specified with four decimal places. The x coordinate SHALL
    // be in the range [0.0001, 0.7400]. The y coordinate SHALL be
    // in the range [0.0001, 0.8400].
    //
    redX:         number,
    redY:         number,
    greenX:       number,
    greenY:       number,
    blueX:        number,
    blueY:        number,
    whitePointX:  number,
    whitePointY:  number,

    //
    // max/min display luminance of the mastering display (in 1 cd/m2 ie. nits)
    //
    // note: ST 2086:2018 - SMPTE Standard specifies minimum display mastering
    // luminance in multiples of 0.0001 cd/m2.
    //
    // For consistency we specify all values
    // in 1 cd/m2. Given that a hypothetical perfect screen has a peak brightness
    // of 10,000 nits and a black level of .0005 nits we do not need to
    // switch units to 0.0001 cd/m2 to increase resolution on the lower end of the
    // minLuminance property. The ranges (in nits) mentioned below suffice
    // the theoretical limit for Mastering Reference Displays and adhere to the
    // SMPTE ST 2084 standard (a.k.a., PQ) which is capable of representing full gamut
    // of luminance level.
    //
    maxLuminance: number,     // [5-10000]
    minLuminance: number,     // [0.0001-5]
  },
}
```

Table: Flag values for the videoFunction property

Function Flag	Usage	Value
SUPPORT_VID_CLIENT_SEEK	Indicates that the client can perform frame-accurate seeks.	0x0001

SUPPORT_VID_CLIENT_HDR	Indicates that the client has support for HDR video. Note: Implies support for colorInfo Object within VideoPacketType.Metadata.	0x0002
SUPPORT_VID_CLIENT_VIDEO_PACKET_TYPE_METADATA	Indicates that the client has support for VideoPacketType.Metadata. See Metadata Frame section for more detail.	0x0004
SUPPORT_VID_CLIENT_LARGE_SCALE_TILE	The large-scale tile allows the decoder to extract only an interesting section in a frame without the need to decompress the entire frame. Support for this feature is not required and is assumed to not be implemented by the client unless this property is present and set to true.	0x0008

Multitrack Streaming via Enhanced RTMP

E-RTMP has introduced support for multitrack streaming, offering increased flexibility in audio and video streaming through the use of a track index (a.k.a., **trackId**). This feature allows for the serialization of multiple tracks over a single [RTMP] connection and stream channel.

It's important to note that multitrack support is designed to augment, not replace, the option of using multiple streams for streaming. While both multiple streams and multitrack can potentially address the same use cases, the choice between them will depend on the specific capabilities of your RTMP implementation and requirements. In certain cases, multitrack may not be the most efficient option.

Sample Multitrack Use Cases

- **Adaptive Bitrate Streaming:** Multitrack support allows the client to send Adaptive Bitrate (ABR) ladders, thus avoiding the need for server-side transcoding and reducing quality loss. This also facilitates sending content with multiple codecs like AV1, HEVC, and VP9.
- **Device Specific Streaming:** The feature allows for the streaming of different aspect ratios, tailored for various device profiles, enabling more dynamic and flexible presentations.
- **Frame-Level Synchronization:** For example, you can synchronize multiple camera views in a concert.
- **Multi-Language Support**: Support for multiple audio tracks in a single [FLV] file is now available, eliminating the need for multiple file versions.

Additional Multitrack Details

- **Video Messages:** Each video message should include a **trackId** (refer to the **videoPacketType.Multitrack** entry in the **ExVideoTagHeader** table within the Enhanced Video section for video bitstream signaling) as it is not persistent across messages.
- **Audio Messages:** Similarly, each audio message should include a **trackId** (refer to the **AudioPacketType.Multitrack** in the **ExAudioTagHeader** table within the Enhanced Audio section for audio bitstream signaling).
- **Payload Parsing:** All tracks within a single timestamp must be processed to ensure comprehensive media handling.
- **Track Ordering:** For identifying the highest priority (a.k.a., default track) or highest quality track, it is RECOMMENDED to use **trackId** set to zero. For tracks of lesser priority or quality, use multiple instances of **trackId** with ascending numerical values. The concept

of **priority** or **quality** can have multiple interpretations, including but not limited to bitrate, resolution, default angle, and language. This recommendation serves as a guideline intended to standardize track numbering across various applications.

General Guidelines

Multitrack capabilities in E-RTMP offer a wide range of possibilities, from adaptive bitrate streaming to multi-language support. While this document doesn't prescribe specific encoding rules or manifest metadata, it aims to guide you through the complexities of leveraging multitrack features. Consider various parameters like codecs, frame rates, key frames, sampling rates, and resolutions to meet your unique objectives. Remember, media encoding settings are separate from E-RTMP configurations.

Enhancing NetConnection connect Command

When a client connects to an E-RTMP server, it sends a **connect** command to the server. The command structure sent from the client to the server contains a Command Object, comprising name-value pairs. This is where the client indicates the audio and video codecs it supports. To declare support for newly defined codecs or other enhancements supported by the client, this name-value pair list must be extended. Below is the description of a new name-value pair used in the Command Object of the **connect** command.

Table: New name-value pair that can be set in the Command Object

Property	Type	Description	Example Value
fourCcList	Strict Array of strings	Used to declare the enhanced list of supported codecs when connecting to the server. The **fourCcList** property is a strict array of dense ordinal indices. Each entry in the array is of string type, specifically a [FourCC] value (i.e., a string that is a sequence of four bytes), representing a supported audio/video codec. In the context of E-RTMP, clients capable of receiving any codec (e.g., recorders or forwarders) may set a **FourCC** value to the wildcard value of '*'. Note: The **fourCcList** property was introduced in the original E-RTMP. Going forward, it is RECOMMENDED on the client side to switch to using the **[audio|video]FourCcInfoMap** properties described below. On the server side, we RECOMMEND supporting both **fourCcList** and **[audio|video]FourCcInfoMap** properties to handle cases where a client has not yet transitioned to using the new properties.	e.g., 1 ['av01', 'vp09', 'hvc1', 'Avc1', 'ac-3', 'ec-3', 'Opus', '.mp3', 'fLaC', 'Aac'] e.g., 2 [*]

videoFourCcInfoMap,
audioFourCcInfoMap
	Object	The **[audio|video]FourCcInfoMap** property is designed to enable setting capability flags for each supported codec in the context of E-RTMP streaming. A **FourCC** key is a four-character code used to specify a video or audio codec. The names of the object properties are strings that correspond to these **FourCC** keys. Each object property holds a numeric value that represents a set of capability flags. These flags can be combined using a Bitwise OR operation.	e.g., 1
			videoFourCcInfoMap = {
			// can forward any video codec
			'*': FourCcInfoMask.CanForward,
			
			// can decode, encode, forward (see '*') VP9 codec
			'vp09': FourCcInfoMask.CanDecode |

		Refer to the enum **FourCcInfoMask** for the available flags:
		
		enum FourCcInfoMask {
		  CanDecode = 0x01,
		  CanEncode = 0x02,
		  CanForward = 0x04,
		}
		
		Capability flags define specific functionalities, such as the ability to decode, encode, or forward.
		
		A FourCC key set to the wildcard character '*' acts as a catch-all for any codec. When this wildcard key exists, it overrides the flags set on properties for specific codecs. For example, if the flag for the '*' property is set to **FourCcInfoMask.CanForward**, all codecs will be forwarded regardless of individual flags set on their specific properties.
			FourCcInfoMask.CanEncode,
			}
			
			e.g., 2
			audioFourCcInfoMap = {
			  // can forward any audio codec
			  '*': FourCcInfoMask.CanForward,
			
			  // can decode, encode, forward (see '*') Opus codec
			  'Opus': FourCcInfoMask.CanDecode |
			  FourCcInfoMask.CanEncode,
			}

capsEx	number	The value represents capability flags which can be combined via a Bitwise OR to indicate which extended set of capabilities (i.e., beyond the legacy RTMP specification) are supported via E-RTMP. See enum CapsExMask for the enumerated values representing the assigned bits. If the extended capabilities are expressed elsewhere they will not appear here (e.g., fourCc, hdr or VideoPacketType.Metadata support is not expressed in this property).
		
		enum CapsExMask {
		  Reconnect = 0x01 // See reconnect section
		  Multitrack = 0x02, // See multitrack section
		}
			CapsExMask.Reconnect | CapsExMask.Multitrack

As you can see, the client declares to the server what enhancements it supports. The server responds with a command, either **_result** or **_error**, to indicate whether the response is a result or an error. During the response, the server provides some properties within an Object as one of the parameters. This is where the server needs to state its support for E-RTMP. The server SHOULD state its support via attributes such as videoFourCcInfoMap, capsEx, and similar properties.

Action Message Format (AMF): AMF0 and AMF3

Action Message Format (AMF) is a compact binary format used to serialize SCRIPTDATA. It has two specifications: [AMF0] and [AMF3]. AMF3 improves on AMF0 by optimizing the payload size on the wire. To understand the full scope of these optimizations, please refer to the AMF0 and AMF3 specifications.

Supporting AMF3 in the [RTMP] and [FLV] is beneficial due to its optimization over AMF0. Understanding the ecosystem is crucial before adding AMF3 support to RTMP or FLV.

Enabling AMF3 in RTMP

To enable support for AMF3 in RTMP, the following steps are REQUIRED:

- Adding support for Data Message, Shared Object Message and Command Message and their associated AMF3 message types (i.e., 15, 16 and 17).
- Adding support for the AMF3 set of possible type markers (see AMF3 specification section 3.1).
- Signaling in the **connect** command that the AMF3 encoding format is supported in addition to AMF0.

RTMP has had AMF3 as part of its specification for some time now. During the handshake, the client declares whether it has support for AMF3.

Enabling AMF3 in FLV

Prior to Y2023, the [FLV] file format did not have AMF3 as part of its SCRIPTDATA specification. To ensure support for AMF3 in FLV:

- Add a new FLV TagType 15 (i.e., in addition to TagType 18), which supports SCRIPTDATA encoded via AMF3 (i.e., similar to the way Data Message is handled).

Important AMF3-encoded Historical Specification Clarification

Established, pre E-RTMP, specifications state the following:

- Command Messages carry the AMF-encoded commands between the client and the server. Message type values:
  - 20 for AMF0 encoding.
  - 17 for AMF3 encoding.
- Data Messages are sent by the client or server to send Metadata or user data to the peer, including details such as creation time, duration, theme, etc. Message type values:
  - 18 for AMF0 encoding.
  - 15 for AMF3 encoding.
- The message types 19 for AMF0 and 16 for AMF3 are reserved for Shared Object events.
- AMF0 was extended to allow an AMF0 encoding context to be switched to AMF3. A new type marker, avmplus-object-marker (byte 0x11), was added. The presence of this marker signifies that the following value is encoded in AMF3. Legacy AMF0 systems that haven't been updated to support AMF3 should throw an unknown type error.

Unfortunately, the above is incomplete and may be somewhat unclear. To clarify, in addition to the above:

- Object Encoding property in the Command Object of the **connect** command indicates the type of serialization (a.k.a., encoding) supported by the client or server:
  - A value of 0 (default and optional) indicates support for AMF0 encoding and message types of 18, 19 and 20.
  - A value of 3 indicates support for both AMF0 and AMF3 encoding and message types of (18, 15), (19, 16) and (20, 17).

- Message payload for message types of 15, 16 and 17 starts with a format selector byte. Currently, only format 0 is defined to indicate AMF0-encoded values. It's possible to signal a switch to AMF3 serialization by prefixing an AMF3 value with an AMF0 avmplus-object-marker (byte 0x11). The switch isn't sticky, and parsing MUST return to AMF0 encoding mode once the AMF3 value is serialized. This means that every AMF3 encoded value MUST be prefixed with an avmplus-object-marker (byte 0x11) as defined in AMF0.

Protocol Versioning

There is no need for a version bump within E-RTMP for either the [RTMP] handshake sequence or the FLV header file version field. All of the enhancements are triggered via the newly defined additions to the bitstream format which don’t break legacy implementations. E-RTMP is self describing in its capabilities.

References

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Appendix

Document Revision History and Guidelines

The revision history section of this document is maintained to provide a clear and concise record of significant changes throughout its development phases, such as alpha, beta, and release. Here are the key points regarding how we manage this history:

- **Phase-Based Documentation:** Important changes made during each phase (alpha, beta, release) are documented in the revision history to keep readers informed of significant developments.
- **Transition Between Phases:** When transitioning from one phase to another (e.g., from alpha to beta), we clear the document revision history. This practice helps keep the document uncluttered and focused on the relevant phase.
- **Exclusion of Minor Changes:** Minor changes that are purely for wording clarification and do not involve adding new features or fixing bugs may be excluded from the revision history.
- **Commit History in GitHub:** The document and its revision history are maintained in GitHub repository at <https://github.com/veovera/enhanced-rtmp>. Although the document revision history is cleared periodically, all commits and their messages are preserved in GitHub, ensuring a comprehensive record of all changes made.
- **Version Changes:** When the version of the specification changes significantly (e.g., from v1 to v2), we again clear the revision history. Despite this, the full history of commits and their messages remains accessible in GitHub.

These guidelines ensure that the revision history in the specification document remains focused, relevant, and easy to navigate, while the complete history of all changes is securely stored and accessible in GitHub.

Table: Revision history

Document Revision History	
Date	Comments
v2-2024-03-16-a1	1. The Enhanced RTMP Version 2 Alpha is now ready for public testing.
v2-2024-04-02-a1	1. Fixed pseudocode logic relating to VP8 sequence start and coded data.
v2-2024-04-22-a1	1. Minor cleanup of normative reference moniker. No changes to logic.
v2-2024-05-9-a1	1. Added audio sections to Enhanced RTMP Version 2 Alpha. 2. Added WebCodecs reference.
v2-2024-05-23-a1	1. Defined the format and behavior of the audio silence message 2. Defined the meaning of AudioPacketType.SequenceEnd 3. Cleaned up the definition for the expected Opus sequence start message 4. Defined the format of the Opus Coded Data on the wire 5. Cleaned up the definition for the expected FLAC sequence start message, it was missing a fLaC marker.
v2-2024-05-24-a1	1. Fixed a bug in pseudocode when parsing FLAC sequence header.
v2-2024-05-28-a1	1. Reworded 'audio silence' message format for more clarity.

v2-2024-05-29-a1	1. Changed page layout to better support .pdf export. No actual spec changes.
v2-...-a*	1. See GitHub for revision history.