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RTP Payload Format for DV (IEC 61834) Video draft-ietf-avt-rfc3189bis-04

Abstract

This document specifies the packetization scheme for encapsulating the compressed digital video data streams commonly known as "DV" into a payload format for the Real-Time Transport Protocol (RTP). This document Obsoletes RFC 3189.

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1. Introduction

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This document specifies payload formats for encapsulating both consumer- and professional-use DV format data streams into the Real-time Transport Protocol (RTP), version 2 [[RFC3550](#)] ([Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," July 2003.](#)). DV compression audio and video formats were designed for a recording format on helical-scan magnetic tape media. The DV standards for consumer-market devices, the IEC 61883 and 61834 series, cover many aspects of consumer-use digital video, including mechanical specifications of a cassette, magnetic recording format, error correction on the magnetic tape, DCT video encoding format, and audio encoding format [[IEC61834](#)] ([IEC, "IEC 61834, Helical-scan digital video cassette recording system using 6,35 mm magnetic tape for consumer use \(525-60, 625-50, 1125-60 and 1250-50 systems\)," .](#)). The digital interface part of IEC 61883 defines an interface on IEEE 1394 system [[IEC61883](#)] ([IEC, "IEC 61883, Consumer audio/video equipment - Digital interface," .](#))[[IEEE1394](#)] ([IEEE, "IEEE Std 1394-1995, Standard for a High Performance Serial Bus," .](#)). This specification set supports several video formats: SD-VCR (Standard Definition), HD-VCR (High Definition), SDL-VCR (Standard Definition - Long), PALPlus, DVB (Digital Video Broadcast) and ATV (Advanced Television). North American formats are indicated with a number of lines and "/60", while European formats use "/50". DV standards extended for professional-use were published by SMPTE as 314M and 370M, for different sampling systems, higher color resolution, and higher bit rates [[SMPTE314M](#)] ([SMPTE, "SMPTE 314M, Data structure for DV-based audio and compressed video 25 and 50Mb/s," .](#))[[SMPTE370M](#)] ([SMPTE, "SMPTE 370M, Data Structure for DV-Based Audio, Data and Compressed Video at 100 Mb/s 1080/60i, 1080/50i, 720/60p, and 720/50p," .](#)).

There are two kinds of DV, one for consumer use and the other for professional. The original "DV" specification designed for consumer-use digital VCRs is approved as the IEC 61834 standard set. The specifications for professional DV are published as SMPTE 314M and 370M. Both encoding formats are based on consumer DV and used in SMPTE D-7, D-9, and D-12 video systems. The RTP payload format specified in this document supports IEC 61834 consumer DV and professional SMPTE 314M and 370M (DV-Based) formats.

IEC 61834 also includes magnetic tape recording for digital TV broadcasting systems (such as DVB and ATV) that use MPEG2 encoding. The payload format for encapsulating MPEG2 into RTP has already been defined in RFC 2250 [[RFC2250](#)] ([Hoffman, D., Fernando, G., Goyal, V., and M. Civanlar, "RTP Payload Format for MPEG1/MPEG2 Video," January 1998.](#)) and others.

Consequently, the payload specified in this document will support six video formats of the IEC standard: SD-VCR (525/60, 625/50), HD-VCR (1125/60, 1250/50) and SDL-VCR (525/60, 625/50), and seven of the SMPTE standards: 314M 25Mbps (525/60, 625/50), 314M 50Mbps (525/60, 625/50), and 370M 100Mbps (1080/60i, 1080/50i, 720/60p, and 720/50p). In the future it can be extended into other video formats managed by 80 byte DV DIF block.

Throughout this specification, we make extensive use of the terminology of IEC and SMPTE standards. The reader should consult the original references for definitions of these terms.

1.1. Terminology

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The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [\[RFC2119\]](#) (Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels," March 1997.).

1.2. The DV Format Encoding

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The DV format only uses the DCT compression technique within each frame, contrasted with the interframe compression of the MPEG video standards [\[ISO/IEC11172\]](#) (ISO/IEC, "ISO/IEC 11172, Coding of moving pictures and associated audio for digital storage media up to about 1,5 Mbits/s.," .) [\[ISO/IEC13818\]](#) (ISO/IEC, "ISO/IEC 13818, Generic coding of moving pictures and associated audio information.," .). All video data, including audio and other system data, are managed within the picture frame unit of video.

The DV video encoding is composed of a three-level hierarchical structure, i.e., DCT super block, DCT macro block, and DCT block. A picture frame is divided into rectangle- or clipped- rectangle-shaped DCT super blocks. DCT super blocks are divided into 27 rectangle- or square-shaped DCT macro blocks, and each DCT macro block consists of a number of DCT blocks. Each DCT block represents rectangle region for each color, Y, Cb, and Cr, and DCT block consists of 8x8 pixels. Audio data is encoded with PCM format. The sampling frequency is 32 kHz, 44.1 kHz or 48 kHz and the quantization is 12-bit non-linear, 16-bit linear or 20-bit linear. The number of channels may be up to 8. Only certain combinations of these parameters are allowed depending upon the video format; the restrictions are specified in each document. A frame of data in the DV format stream is divided into several "DIF sequences". A DIF sequence is composed of an integral number of 80- byte DIF blocks. A DIF block is the primitive unit for all treatment of DV

streams. Each DIF block contains a 3-byte ID header that specifies the type of the DIF block and its position in the DIF sequence. Five types of DIF blocks are defined: DIF sequence header, Subcode, Video Auxiliary information (VAUX), Audio, and Video. Audio DIF blocks are composed of 5 bytes of Audio Auxiliary data (AAUX) and 72 bytes of audio data. Each RTP packet starts with the RTP header as defined in RFC 3550 [\[RFC3550\] \(Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," July 2003.\)](#). No additional payload-format-specific header is required for this payload format.

2. RTP Payload Format

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2.1. RTP Header Usage

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The RTP header fields that have a meaning specific to the DV format are described as follows:

Payload type (PT): The payload type is dynamically assigned by means outside the scope of this document. If multiple DV encoding formats are to be used within one RTP session, then multiple dynamic payload types **MUST** be assigned, one for each DV encoding format. The sender **MUST** change to the corresponding payload type whenever the encoding format is changed.

Timestamp: 32-bit 90 kHz timestamp representing the time at which the first data in the frame was sampled. All RTP packets within the same video frame **MUST** have the same timestamp. The timestamp **SHOULD** increment by a multiple of the nominal interval for one DV frame time, as given in the following table:

Mode	Frame rate (Hz)	Increase of one DV frame in 90kHz timestamp
525-60	29.97	3003
625-50	25	3600
1125-60	30	3000
1250-50	25	3600
1080-60i	29.97	3003
1080-50i	25	3600
720-60p	59.94	3003(*)
720-50p	50	3600(*)

Note that even in 720-line DV system, the data in two video frame shall be processed within one DV frame duration of 1080-line system. Audio data and subcode data in 720-line system are processed in the same way as the 1080-line system. Therefore in 720-line system, the increase of one DV frame corresponds two video frames time.

Marker bit (M): The marker bit of the RTP fixed header is set to one on the last packet of a video frame, and otherwise, must be zero. The M bit allows the receiver to know that it has received the last packet of a frame so it can display the image without waiting for the first packet of the next frame to arrive to detect the frame change. However, detection of a frame change MUST NOT rely on the marker bit since the last packet of the frame might be lost. Detection of a frame change MUST be based on a difference in the RTP timestamp.

2.2. Payload Structures

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Integral DIF blocks are placed into the RTP payload beginning immediately after the RTP header. Any number of DIF blocks may be packed into one RTP packet, except that all DIF blocks in one RTP packet MUST be from the same video frame. DIF blocks from the next video frame MUST NOT be packed into the same RTP packet even if more payload space remains. This requirement stems from the fact that the transition from one video frame to the next is indicated by a change in the RTP timestamp. It also reduces the processing complexity on the receiver. Since the RTP payload contains an integral number of DIF blocks, the length of the RTP payload will be a multiple of 80 bytes.

Audio and video data may be transmitted as one bundled RTP stream or in separate RTP streams (unbundled). The choice MUST be indicated as part of the assignment of the dynamic payload type and MUST remain unchanged for the duration of the RTP session to avoid complicated procedures of sequence number synchronization. The RTP sender could omit DIF-sequence header and subcode DIF blocks from a stream, in the case of the information either is known out-of-band or is not be required for the application. Note that time code in DIF blocks is mandatory for professional video applications. When sending DIF- sequence header and subcode DIF blocks with unbundled audio and video streams, both types of blocks MUST be included in the video stream.

DV streams include "source" and "source control" packs that carry information indispensable for proper decoding, such as video signal type, frame rate, aspect ratio, picture position, quantization of audio sampling, number of audio samples in a frame, number of audio channels, audio channel assignment, and language of the audio. However, describing all of these attributes with a signaling protocol would require large descriptions to enumerate all the combinations. Therefore, no Session Description Protocol (SDP) [\[RFC4566\] \(Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol," July 2006.\)](#) parameters for

these attributes are defined in this document. Instead, the RTP sender MUST transmit at least those VAUX DIF blocks and/or audio DIF blocks with AAUX information bytes that include "source" and "source control" packs containing the indispensable information for decoding.

In the case of one bundled stream, DIF blocks for both audio and video are packed into RTP packets in the same order as they were encoded.

In the case of an unbundled stream, only the header, subcode, video and VAUX DIF blocks are sent within the video stream. Audio is sent in a different stream if desired, using a different RTP payload type. It is also possible to send audio duplicated in a separate stream, in addition to bundling it in with the video stream.

When using unbundled mode, it is RECOMMENDED that the audio stream data be extracted from the DIF blocks and repackaged into the corresponding RTP payload format for the audio encoding (DAT12, L16, L20) [[RFC3551](#)] ([Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control," July 2003.](#))[[RFC3190](#)] ([Kobayashi, K., Ogawa, A., Casner, S., and C. Bormann, "RTP Payload Format for 12-bit DAT Audio and 20- and 24-bit Linear Sampled Audio," January 2002.](#)) in order to maximize interoperability with non-DV- capable receivers while maintaining the original source quality.

In the case of unbundled transmission where both audio and video are sent in the DV format, the same timestamp SHOULD be used for both audio and video data within the same frame to simplify the lip synchronization effort on the receiver. Lip synchronization may also be achieved using reference timestamps passed in RTCP as described in RFC 3550.

The sender MAY reduce the video frame rate by discarding the video data and VAUX DIF blocks for some of the video frames. The RTP timestamp MUST still be incremented to account for the discarded frames. The sender MAY alternatively reduce bandwidth by discarding video data DIF blocks for portions of the image which are unchanged from the previous image. To enable this bandwidth reduction, receivers SHOULD implement an error concealment strategy to accommodate lost or missing DIF blocks, e.g., repeating the corresponding DIF block from the previous image.

3. Payload Format Parameters

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This section specifies the parameters that MAY be used to select optional features of the payload format and certain features of the bitstream. The parameters are specified here as part of the media type registration for the DV encoding. A mapping of the parameters into the Session Description Protocol (SDP) [[RFC4566](#)] ([Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol," July 2006.](#)) is also provided for applications that use SDP. Equivalent parameters could be defined elsewhere for use with control protocols that do not use SDP.

3.1. Media Type Registration

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This registration is done using the template defined in RFC 4288 [[RFC4288](#)] (Freed, N. and J. Klensin, "Media Type Specifications and Registration Procedures," December 2005.) and following RFC 4855 [[RFC4855](#)] (Casner, S., "Media Type Registration of RTP Payload Formats," February 2007.).

3.1.1. Media Type Registration for DV Video

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Type name: video

Subtype name: DV

Required parameters:

encode: type of DV format. Permissible values for encode are SD-VCR/525-60, SD-VCR/625-50, HD-VCR/1125-60 HD-VCR/1250-50, SDL-VCR/525-60, SDL-VCR/625-50, 314M-25/525-60, 314M-25/625-50, 314M-50/525-60, 314M-50/625-50, 370M/1080-60i, 370M/1080-50i, 370M/720-60p, 370M/720-50p, 306M/525-60 (for backward compatibility), and 306M/625-50 (for backward compatibility).

Optional parameters:

audio: whether the DV stream includes audio data or not. Permissible values for audio are bundled and none. Defaults to none.

Encoding considerations: DV video can be transmitted with RTP as specified in RFCXXXX (This document). Other transport methods are not specified.

Security considerations: See [Section 4 \(Security Considerations\)](#) of RFCXXXX (This document).

Interoperability considerations: NONE

Public specification: IEC 61834 Standard

SMPTE 314M

SMPTE 370M

RFCXXXX (This document)

SMPTE 314M (for backward compatibility).

Applications that use this media type: Audio and video streaming and conferencing tools.

Additional information: NONE

Person & email address to contact for further information: Katsushi Kobayashi
e-mail: ikob@ni.aist.go.jp

Intended usage: COMMON

Restrictions on usage: This media type depends on RTP framing, and hence is only defined for transfer via RTP (RFC 3550). Transfer within other framing protocols is not defined at this time.

Author: Katsushi Kobayashi

Change controller: IETF Audio/Video Transport working group delegated from the IESG

3.1.2. Media Type Registration for DV Audio

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Type name: audio

Subtype name: DV

Required parameters:

encode: type of DV format. Permissible values for encode are SD-VCR/525-60, SD-VCR/625-50, HD-VCR/1125-60 HD-VCR/1250-50, SDL-VCR/525-60, SDL-VCR/625-50, 314M-25/525-60, 314M-25/625-50, 314M-50/525-60, 314M-50/625-50, 370M/1080-60i, 370M/1080-50i, 370M/720-60p, 370M/720-50p, 306M/

525-60 (for backward compatibility), and 306M/625-50 (for backward compatibility).

Optional parameters:

audio: whether the DV stream includes audio data or not. Permissible values for audio are bundled and none. Defaults to none.

Encoding considerations: DV video can be transmitted with RTP as specified in RFCXXXX (This document). Other transport methods are not specified.

Security considerations: See [Section 4 \(Security Considerations\)](#) of RFCXXXX (This document).

Interoperability considerations: NONE

Published specification: IEC 61834 Standard

SMPTE 314M

SMPTE 370M

RFCXXXX (This document)

SMPTE 314M (for backward compatibility).

Applications that use this media type: Audio and video streaming and conferencing tools.

Additional information: NONE

Person & email address to contact for further information: Katsushi Kobayashi

e-mail: ikob@ni.aist.go.jp

Intended usage: COMMON

Restrictions on usage: This media type depends on RTP framing, and hence is only defined for transfer via RTP (RFC 3550). Transfer within other framing protocols is not defined at this time.

Author: Katsushi Kobayashi

Change controller: IETF Audio/Video Transport working group delegated from the IESG

3.2. SDP Parameters

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3.2.1. Mapping of Payload Type Parameters to SDP

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The information carried in the media type specification has a specific mapping to fields in the Session Description Protocol (SDP), which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing the DV encoding, the mapping is as follows:

*The media type ("video") goes in SDP "m=" as the media name.

*The media subtype ("DV") goes in SDP "a=rtpmap" as the encoding name. The RTP clock rate in "a=rtpmap" MUST be 90000 which for the payload format defined in this document is a 90kHz clock.

*Any remaining parameters go in the SDP "a=fmtp" attribute by copying them directly from the media type string as a semicolon separated list of parameter=value pairs.

Note that the examples in RFC3189 (older version of this document) are incorrect on the SDP "a=fmtp" attribute describing.

In the DV video payload format, the a=fmtp line will be used to show the encoding type within the DV video and will be used as below:

```
a=fmtp:<payload type> encode=<DV-video encoding>
```

The required parameter <DV-video encoding> specifies which type of DV format is used. The DV format name will be one of the following:

```
SD-VCR/525-60
```

```
SD-VCR/625-50
```

```
HD-VCR/1125-60
```

```
HD-VCR/1250-50
```

```
SDL-VCR/525-60
```

```
SDL-VCR/625-50
```

```
314M-25/525-60
```

```
314M-25/625-50
```

```
314M-50/525-60
```

314M-50/625-50
370M/1080-60i
370M/1080-50i
370M/720-60p
370M/720-50p
306M/525-60 (for backward compatibility)
306M/625-50 (for backward compatibility)

In order to show whether the audio data is bundled into the DV stream or not, a format specific parameter is defined as below:

```
a=fmtp:<payload type> audio=<audio bundled>
```

The optional parameter <audio bundled> will be one of the following:

```
bundled  
none (default)
```

If the fmtp audio parameter is not present, then audio data MUST NOT be bundled into the DV video stream.

3.2.2. Usage with the SDP Offer/Answer Model

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The following considerations apply when using SDP offer-answer procedures [\[RFC4566\] \(Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol," July 2006.\)](#) to negotiate the use of DV payload in RTP:

*The "encode" parameter can be used for bi-directional, mono-directional and multicast streams. If the offerer sets a encode type on a=fmtp field, the answerer MUST select one encode type, and reply with selected encode type value.

*Any unknown parameter in an offer MUST be ignored by the receiver and MUST NOT be included in the answer.

Some special rules apply for mono-directional traffic:

*The optional "audio" parameter is only used for the bundled stream. On the offerer sets a audio bundled type on a=fmtp field,

then the answerer MUST select whether the DV stream should be included audio data or not, and reply with selected value.

Some special rules apply for multicast:

*The "encode" and "audio" parameter becomes declarative and MUST NOT be negotiated. This parameter is fixed, and a participant MUST use the configuration that is provided for the session.

3.3. Examples

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Some example SDP session descriptions utilizing DV encoding formats follow.

3.3.1. Example for Unbundled Streams

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When using unbundled mode, the RTP streams for video and audio will be sent separately to different ports or different multicast groups. When this is done, SDP carries several m=?? lines, one for each media type of the session (see RFC 4566).

An example SDP description using these attributes is:

```
v=0
o=ikob 2890844526 2890842807 IN IP4 192.0.2.1
s=POI Seminar
i=A Seminar on how to make Presentations on the Internet
u=http://www.example.net/~ikob/POI/index.html
e=ikob@example.net (Katsushi Kobayashi)
c=IN IP4 233.252.0.1/127
t=2873397496 2873404696
m=audio 49170 RTP/AVP 112
a=rtpmap:112 L16/32000/2
m=video 50000 RTP/AVP 113
a=rtpmap:113 DV/90000
a=fmtp:113 encode=SD-VCR/525-60; audio=none
```

This describes a session where audio and video streams are sent separately. The session is sent to a multicast group 233.252.0.1. The audio is sent using L16 format, and the video is sent using SD- VCR 525/60 format which corresponds to NTSC format in consumer DV.

3.3.2. Example for Bundled Streams

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When sending a bundled stream, all the DIF blocks including system data will be sent through a single RTP stream.

An example SDP description for a bundled DV stream is:

```
v=0
o=ikob 2890844526 2890842807 IN IP4 192.0.2.1
s=POI Seminar
i=A Seminar on how to make Presentations on the Internet
u=http://www.example.net/~ikob/POI/index.html
e=ikob@example.net (Katsushi Kobayashi)
c=IN IP4 233.252.0.1/127
t=2873397496 2873404696
m=video 49170 RTP/AVP 112 113
a=rtpmap:112 DV/90000
a=fmtp: 112 encode=SD-VCR/525-60; audio=bundled
a=fmtp: 113 encode=314M-50/525-60; audio=bundled
```

This SDP record describes a session where audio and video streams are sent bundled. The session is sent to a multicast group 233.252.0.1. The video is sent using both 525/60 consumer DV and SMPTE standard 314M 50Mbps formats, when the payload type is 112 and 113, respectively.

4. Security Considerations

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RTP packets using the payload format defined in this specification are subject to the security considerations discussed in the RTP specification [\[RFC3550\] \(Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," July 2003.\)](#), and any appropriate RTP profile. This implies that confidentiality of the media streams is achieved by encryption. Because the data compression used with this payload format is applied to end-to-end, encryption may be performed after compression so there is no conflict between the two operations.

A potential denial-of-service threat exists for data encodings using compression techniques that have non-uniform receiver-end computational load. The attacker can inject pathological datagrams into the stream which are complex to decode and cause the receiver to be overloaded. However, this encoding does not exhibit any significant non-uniformity. As with any IP-based protocol, in some circumstances a receiver may be overloaded simply by the receipt of too many packets, either desired or undesired. Network-layer authentication may be used to discard packets from undesired sources, but the processing cost of the authentication itself may be too high. In a multicast environment, joining and pruning mechanism of specific sources is specified in IGMPv3 and MLDv2 [\[RFC3376\]](#)

[\(Cain, B., Deering, S., Kouvelas, I., Fenner, B., and A. Thyagarajan, "Internet Group Management Protocol, Version 3," October 2002.\)](#)[\[RFC3810\]](#)
[\(Vida, R. and L. Costa, "Multicast Listener Discovery Version 2 \(MLDv2\) for IPv6," June 2004.\)](#) and in multicast routing protocols to allow a receiver to select which sources are allowed to reach it [\[RFC4607\]](#)
[\(Holbrook, H. and B. Cain, "Source-Specific Multicast for IP," August 2006.\)](#).

5. Congestion Control

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The general congestion control considerations for transporting RTP data apply; see RTP [\[RFC3550\]](#) [\(Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," July 2003.\)](#) and any applicable RTP profile like AVP [\[RFC3551\]](#) [\(Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control," July 2003.\)](#).

The number of frames encapsulated in each RTP payload highly influences the overall bandwidth of the RTP stream due to header overhead constraints. Packetizing more frames in each RTP payload can reduce the number of packets sent and hence the header overhead, at the expense of increased delay and reduced error robustness. If forward error correction (FEC) is used, the amount of FEC-induced redundancy needs to be regulated such that the use of FEC itself does not cause a congestion problem.

6. IANA Considerations

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This document defines a new RTP payload name and associated Media Type, DV. The registration forms (based on the RFC 4855 [\[RFC4855\]](#) [\(Casner, S., "Media Type Registration of RTP Payload Formats," February 2007.\)](#) definition) for the Media Types for both video and audio are shown in [Section 3.1 \(Media Type Registration\)](#).

7. Major Changes from RFC 3189

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The changes from RFC 3189 are:

1. Removed SMPTE 306M, since it can covered SMPTE 314M format.
2. Added SMPTE 370M 100Mbps HDTV (1080/60i, 1080/50i, 720/60p, and 720/50p) format.

3. Incorporated Source Specific Multicast (SSM) spec. for avoiding overloaded traffic source in multicast usage.
 4. Clarified the case that the sender omit subcode DIF block data from the stream.
 5. Added the Offer-Answer Model Consideration.
 6. Revised Media Types registration form based on new registration rule (RFC 4855).
 7. Revised section structure from based on new example.
-

8. Interoperability with Previous Implementations

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In this section, we will specify the interoperability issue with the implementations based on RFC obsoleted by this document.

RFC 3189 regards SMPTE306M [[SMPTE306M](#)] ([SMPTE, "SMPTE 306M, 6.35-mm type D-7 component format - video compression at 25Mb/s -525/60 and 625/50.,"](#) [.](#)) and SMPTE314M as different encoding format, although the format of SMPTE 306M is already covered SMPTE 314M. Therefore, this document recommends that the definition depending on SMPTE306M SHOULD NOT be used, and SHOULD use SMPTE314M instead. An RTP application could handle a stream identified as SMPTE306M encoding as SMPTE314M one.

9. References

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9.1. Normative References

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