原 FFmpeg源代码简单分析：av_write_frame()

===

FFmpeg的库函数源代码分析文章列表：

【架构图】

FFmpeg 源代码结构图 - 解码

FFmpeg 源代码结构图 - 编码

【通用】

FFmpeg 源代码简单分析： av_register_all()

FFmpeg 源代码简单分析： avcodec_register_all()

FFmpeg 源代码简单分析：内存的分配和释放（ av_malloc() 、 av_free() 等）

FFmpeg 源代码简单分析：常见结构体的初始化和销毁（ AVFormatContext ， AVFrame 等）

FFmpeg 源代码简单分析： avio_open2()

FFmpeg 源代码简单分析： av_find_decoder() 和 av_find_encoder()

FFmpeg 源代码简单分析： avcodec_open2()

FFmpeg 源代码简单分析： avcodec_close()

【解码】

图解 FFMPEG 打开媒体的函数 avformat_open_input

FFmpeg 源代码简单分析： avformat_open_input()

FFmpeg 源代码简单分析： avformat_find_stream_info()

FFmpeg 源代码简单分析： av_read_frame()

FFmpeg 源代码简单分析： avcodec_decode_video2()

FFmpeg 源代码简单分析： avformat_close_input()

【编码】

FFmpeg 源代码简单分析： avformat_alloc_output_context2()

FFmpeg 源代码简单分析： avformat_write_header()

FFmpeg 源代码简单分析： avcodec_encode_video()

FFmpeg 源代码简单分析： av_write_frame()

FFmpeg 源代码简单分析： av_write_trailer()

【其它】

FFmpeg 源代码简单分析：日志输出系统（ av_log() 等）

FFmpeg 源代码简单分析：结构体成员管理系统 -AVClass

FFmpeg 源代码简单分析：结构体成员管理系统 -AVOption

FFmpeg 源代码简单分析： libswscale 的 sws_getContext()

FFmpeg 源代码简单分析： libswscale 的 sws_scale()

FFmpeg 源代码简单分析： libavdevice 的 avdevice_register_all()

FFmpeg 源代码简单分析： libavdevice 的 gdigrab

【脚本】

FFmpeg 源代码简单分析： makefile

FFmpeg 源代码简单分析： configure

【H.264】

FFmpeg 的 H.264 解码器源代码简单分析：概述

===

打算写两篇文章简单分析FFmpeg的写文件用到的3个函数avformat_write_header()，av_write_frame()以及av_write_trailer()。上篇文章已经分析了avformat_write_header()，这篇文章继续分析av_write_frame()。

av_write_frame()用于输出一帧视音频数据，它的声明位于libavformat\avformat.h，如下所示。

```cpp
/**
 * Write a packet to an output media file.
 *
 * This function passes the packet directly to the muxer, without any buffering
 * or reordering. The caller is responsible for correctly interleaving the
 * packets if the format requires it. Callers that want libavformat to handle
 * the interleaving should call av_interleaved_write_frame() instead of this
 * function.
 *
 * @param s media file handle
 * @param pkt The packet containing the data to be written. Note that unlike
 *            av_interleaved_write_frame(), this function does not take
 *            ownership of the packet passed to it (though some muxers may make
 *            an internal reference to the input packet).
 *            <br>
 *            This parameter can be NULL (at any time, not just at the end), in
 *            order to immediately flush data buffered within the muxer, for
 *            muxers that buffer up data internally before writing it to the
 *            output.
 *            <br>
 *            Packet's @ref AVPacket.stream_index "stream_index" field must be
 *            set to the index of the corresponding stream in @ref
 *            AVFormatContext.streams "s->streams". It is very strongly
 *            recommended that timing information (@ref AVPacket.pts "pts", @ref
 *            AVPacket.dts "dts", @ref AVPacket.duration "duration") is set to
 *            correct values.
 * @return < 0 on error, = 0 if OK, 1 if flushed and there is no more data to flush
 *
 * @see av_interleaved_write_frame()
 */
int av_write_frame(AVFormatContext *s, AVPacket *pkt);
```

简单解释一下它的参数的含义：

s：用于输出的AVFormatContext。

pkt：等待输出的AVPacket。

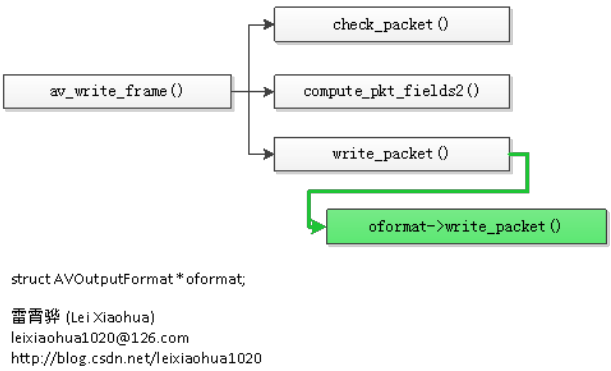
函数正常执行后返回值等于0。

这个函数最典型的例子可以参考：

最简单的基于FFMPEG的视频编码器（YUV编码为H.264）

函数调用关系图

av_write_frame()的调用关系如下图所示。



av_write_frame()

av_write_frame()的定义位于libavformat\mux.c，如下所示。

```cpp
int av_write_frame(AVFormatContext *s, AVPacket *pkt)
{
    int ret;

    ret = check_packet(s, pkt);
    if (ret < 0)
        return ret;
    //Packet为NULL，Flush Encoder
    if (!pkt) {
        if (s->oformat->flags & AVFMT_ALLOW_FLUSH) {
            ret = s->oformat->write_packet(s, NULL);
            if (s->flush_packets && s->pb && s->pb->error >= 0 && s->flags & AVFMT_FLAG_FLUSH_PACKETS)
                avio_flush(s->pb);
            if (ret >= 0 && s->pb && s->pb->error < 0)
                ret = s->pb->error;
            return ret;
        }
        return 1;
    }

    ret = compute_pkt_fields2(s, s->streams[pkt->stream_index], pkt);

    if (ret < 0 && !(s->oformat->flags & AVFMT_NOTIMESTAMPS))
        return ret;
    //写入
    ret = write_packet(s, pkt);
    if (ret >= 0 && s->pb && s->pb->error < 0)
        ret = s->pb->error;

    if (ret >= 0)
        s->streams[pkt->stream_index]->nb_frames++;
    return ret;
}
```

从源代码可以看出，av_write_frame()主要完成了以下几步工作：
（1）调用check_packet()做一些简单的检测
（2）调用compute_pkt_fields2()设置AVPacket的一些属性值
（3）调用write_packet()写入数据
下面分别看一下这几个函数功能。

check_packet()

check_packet()定义位于libavformat\mux.c，如下所示。

```cpp
static int check_packet(AVFormatContext *s, AVPacket *pkt)
{
    if (!pkt)
        return 0;

    if (pkt->stream_index < 0 || pkt->stream_index >= s->nb_streams) {
        av_log(s, AV_LOG_ERROR, "Invalid packet stream index: %d\n",
               pkt->stream_index);
        return AVERROR(EINVAL);
    }

    if (s->streams[pkt->stream_index]->codec->codec_type == AVMEDIA_TYPE_ATTACHMENT) {
        av_log(s, AV_LOG_ERROR, "Received a packet for an attachment stream.\n");
        return AVERROR(EINVAL);
    }

    return 0;
}
```

从代码中可以看出，check_packet()的功能比较简单：首先检查一下输入的AVPacket是否为空，如果为空，则是直接返回；然后检查一下AVPacket的stream_index（标记了该AVPacket所属的AVStream）设置是否正常，如果为负数或者大于AVStream的个数，则返回错误信息；最后检查AVPacket所属的AVStream是否属于attachment stream，这个地方没见过，目前还没有研究。

compute_pkt_fields2()

compute_pkt_fields2()函数的定义位于libavformat\mux.c，如下所示。

```cpp
//FIXME merge with compute_pkt_fields
```

```c
static int compute_pkt_fields2(AVFormatContext *s, AVStream *st, AVPacket *pkt)
{
    int delay = FFMAX(st->codec->has_b_frames, st->codec->max_b_frames > 0);
    int num, den, i;
    int frame_size;

    av_dlog(s, "compute_pkt_fields2: pts:%s dts:%s cur_dts:%s b:%d size:%d st:%d\n",
            av_ts2str(pkt->pts), av_ts2str(pkt->dts), av_ts2str(st->cur_dts), delay, pkt->size, pkt->stream_index);

    if (pkt->duration < 0 && st->codec->codec_type != AVMEDIA_TYPE_SUBTITLE) {
        av_log(s, AV_LOG_WARNING, "Packet with invalid duration %d in stream %d\n",
               pkt->duration, pkt->stream_index);
        pkt->duration = 0;
    }

    /* duration field */
    if (pkt->duration == 0) {
        ff_compute_frame_duration(s, &num, &den, st, NULL, pkt);
        if (den && num) {
            pkt->duration = av_rescale(1, num * (int64_t)st->time_base.den * st->codec->ticks_per_frame, den * (int64_t)st->time_base.num);
        }
    }

    if (pkt->pts == AV_NOPTS_VALUE && pkt->dts != AV_NOPTS_VALUE && delay == 0)
        pkt->pts = pkt->dts;

    //XXX/FIXME this is a temporary hack until all encoders output pts
    if ((pkt->pts == 0 || pkt->pts == AV_NOPTS_VALUE) && pkt->dts == AV_NOPTS_VALUE && !delay) {
        static int warned;
        if (!warned) {
            av_log(s, AV_LOG_WARNING, "Encoder did not produce proper pts, making some up.\n");
            warned = 1;
        }
        pkt->dts =
//        pkt->pts= st->cur_dts;
            pkt->pts = st->pts.val;
    }

    //calculate dts from pts
    if (pkt->pts != AV_NOPTS_VALUE && pkt->dts == AV_NOPTS_VALUE && delay <= MAX_REORDER_DELAY) {
        st->pts_buffer[0] = pkt->pts;
        for (i = 1; i < delay + 1 && st->pts_buffer[i] == AV_NOPTS_VALUE; i++)
            st->pts_buffer[i] = pkt->pts + (i - delay - 1) * pkt->duration;
        for (i = 0; i<delay && st->pts_buffer[i] > st->pts_buffer[i + 1]; i++)
            FFSWAP(int64_t, st->pts_buffer[i], st->pts_buffer[i + 1]);

        pkt->dts = st->pts_buffer[0];
    }

    if (st->cur_dts && st->cur_dts != AV_NOPTS_VALUE &&
        ((!(s->oformat->flags & AVFMT_TS_NONSTRICT) &&
          st->cur_dts >= pkt->dts) || st->cur_dts > pkt->dts)) {
        av_log(s, AV_LOG_ERROR,
               "Application provided invalid, non monotonically increasing dts to muxer in stream %d: %s >= %s\n",
               st->index, av_ts2str(st->cur_dts), av_ts2str(pkt->dts));
        return AVERROR(EINVAL);
    }
    if (pkt->dts != AV_NOPTS_VALUE && pkt->pts != AV_NOPTS_VALUE && pkt->pts < pkt->dts) {
        av_log(s, AV_LOG_ERROR,
               "pts (%s) < dts (%s) in stream %d\n",
               av_ts2str(pkt->pts), av_ts2str(pkt->dts),
               st->index);
        return AVERROR(EINVAL);
    }

    av_dlog(s, "av_write_frame: pts2:%s dts2:%s\n",
            av_ts2str(pkt->pts), av_ts2str(pkt->dts));
    st->cur_dts = pkt->dts;
    st->pts.val = pkt->dts;

    /* update pts */
    switch (st->codec->codec_type) {
    case AVMEDIA_TYPE_AUDIO:
        frame_size = (pkt->flags & AV_PKT_FLAG_UNCODED_FRAME) ?
                     ((AVFrame *)pkt->data)->nb_samples :
                     av_get_audio_frame_duration(st->codec, pkt->size);

        /* HACK/FIXME, we skip the initial 0 size packets as they are most
         * likely equal to the encoder delay, but it would be better if we
         * had the real timestamps from the encoder */
        if (frame_size >= 0 && (pkt->size || st->pts.num != st->pts.den >> 1 || st->pts.val)) {
            frac_add(&st->pts, (int64_t)st->time_base.den * frame_size);
        }
        break;
    case AVMEDIA_TYPE_VIDEO:
        frac_add(&st->pts, (int64_t)st->time_base.den * st->codec->time_base.num);
        break;
    }
    return 0;
}
```

从代码中可以看出，compute_pkt_fields2()主要有两方面的功能：一方面用于计算AVPacket的duration， dts等信息；另一方面用于检查pts、dts这些参数的合理性（例如PTS是否一定大于DTS）。具体的代码还没有细看，以后有时间再进行分析。

AVOutputFormat->write_packet()

write_packet()函数的定义位于libavformat\mux.c，如下所示。

[cpp]

```cpp
/**
 * Make timestamps non negative, move side data from payload to internal struct, call muxer, and restore
 * sidedata.
 *
 * FIXME: this function should NEVER get undefined pts/dts beside when the
 * AVFMT_NOTIMESTAMPS is set.
 * Those additional safety checks should be dropped once the correct checks
 * are set in the callers.
 */
static int write_packet(AVFormatContext *s, AVPacket *pkt)
{
    int ret, did_split;

    if (s->output_ts_offset) {
        AVStream *st = s->streams[pkt->stream_index];
        int64_t offset = av_rescale_q(s->output_ts_offset, AV_TIME_BASE_Q, st->time_base);

        if (pkt->dts != AV_NOPTS_VALUE)
            pkt->dts += offset;
        if (pkt->pts != AV_NOPTS_VALUE)
            pkt->pts += offset;
    }

    if (s->avoid_negative_ts > 0) {
        AVStream *st = s->streams[pkt->stream_index];
        int64_t offset = st->mux_ts_offset;

        if (s->offset == AV_NOPTS_VALUE && pkt->dts != AV_NOPTS_VALUE &&
            (pkt->dts < 0 || s->avoid_negative_ts == AVFMT_AVOID_NEG_TS_MAKE_ZERO)) {
            s->offset = -pkt->dts;
            s->offset_timebase = st->time_base;
        }

        if (s->offset != AV_NOPTS_VALUE && !offset) {
            offset = st->mux_ts_offset =
                av_rescale_q_rnd(s->offset,
                                 s->offset_timebase,
                                 st->time_base,
                                 AV_ROUND_UP);
        }

        if (pkt->dts != AV_NOPTS_VALUE)
            pkt->dts += offset;
        if (pkt->pts != AV_NOPTS_VALUE)
            pkt->pts += offset;

        av_assert2(pkt->dts == AV_NOPTS_VALUE || pkt->dts >= 0 || s->max_interleave_delta > 0);
        if (pkt->dts != AV_NOPTS_VALUE && pkt->dts < 0) {
            av_log(s, AV_LOG_WARNING,
                   "Packets poorly interleaved, failed to avoid negative "
                   "timestamp %s in stream %d.\n"
                   "Try -max_interleave_delta 0 as a possible workaround.\n",
                   av_ts2str(pkt->dts),
                   pkt->stream_index
            );
        }
    }

    did_split = av_packet_split_side_data(pkt);
    if ((pkt->flags & AV_PKT_FLAG_UNCODED_FRAME)) {
        AVFrame *frame = (AVFrame *)pkt->data;
        av_assert0(pkt->size == UNCODED_FRAME_PACKET_SIZE);
        ret = s->oformat->write_uncoded_frame(s, pkt->stream_index, &frame, 0);
        av_frame_free(&frame);
    } else {
        //写入
        ret = s->oformat->write_packet(s, pkt);
    }

    if (s->flush_packets && s->pb && ret >= 0 && s->flags & AVFMT_FLAG_FLUSH_PACKETS)
        avio_flush(s->pb);

    if (did_split)
        av_packet_merge_side_data(pkt);

    return ret;
}
```

write_packet()函数最关键的地方就是调用了AVOutputFormat中写入数据的方法。如果AVPacket中的flag标记中包含AV_PKT_FLAG_UNCODED_FRAME，就会调用AVOutputFormat的write_uncoded_frame()函数；如果不包含那个标记，就会调用write_packet()函数。write_packet()实际上是一个函数指针，指向特定的AVOutputFormat中的实现函数。例如，我们看一下FLV对应的AVOutputFormat，位于libavformat\flvenc.c，如下所示。

```cpp
AVOutputFormat ff_flv_muxer = {
    .name           = "flv",
    .long_name      = NULL_IF_CONFIG_SMALL("FLV (Flash Video)"),
    .mime_type      = "video/x-flv",
    .extensions     = "flv",
    .priv_data_size = sizeof(FLVContext),
    .audio_codec    = CONFIG_LIBMP3LAME ? AV_CODEC_ID_MP3 : AV_CODEC_ID_ADPCM_SWF,
    .video_codec    = AV_CODEC_ID_FLV1,
    .write_header   = flv_write_header,
    .write_packet   = flv_write_packet,
    .write_trailer  = flv_write_trailer,
    .codec_tag      = (const AVCodecTag* const []) {
                          flv_video_codec_ids, flv_audio_codec_ids, 0
                      },
    .flags          = AVFMT_GLOBALHEADER | AVFMT_VARIABLE_FPS |
                      AVFMT_TS_NONSTRICT,
};
```

从ff_flv_muxer的定义可以看出，write_packet()指向的是flv_write_packet()函数。在看flv_write_packet()函数的定义之前，我们先回顾一下FLV封装格式的结构。

FLV封装格式

FLV封装格式如下图所示。

PS：原图是网上找的，感觉画的很清晰，比官方的Video File Format Specification更加通俗易懂。但是图中有一个错误，就是TagHeader中的StreamID字段的长度写错了（查看了一下官方标准，应该是3字节，现在已经改过来了）。

<table>
<tr><td rowspan="4">Flv Header</td><td colspan="3">Signature（3 字节）为文件标识，总为"FLV"，（0x46，0x4c，0x66）</td></tr>
<tr><td colspan="3">Version（1 字节）为版本，目前为 0x01</td></tr>
<tr><td colspan="3">Flags（1 字节）前 5 位保留，必须为 0。第 6 位表示是否存在音频 Tag。第 7 位保留，必须为 0。第 8 位表示是否存在视频 Tag。</td></tr>
<tr><td colspan="3">Headersize（4 字节）为从 File Header 开始到 File Body 开始的字节数，版本 1 中总为 9。</td></tr>
<tr><td rowspan="13">Flv Body</td><td colspan="3">Previous Tag Size #0（4 字节）表示前一个 Tag 的长度</td></tr>
<tr><td rowspan="6">Tag #1</td><td rowspan="5">Tag Header</td><td>Type（1 字节）表示 Tag 类型，包括音频（0x08），视频（0x09）和 script data（0x12），其他类型值被保留</td></tr>
<tr><td>Datasize（3 字节）表示该 Tag Ddata 部分的大小</td></tr>
<tr><td>Timestamp（3 字节）表示该 Tag 的时间戳</td></tr>
<tr><td>Timestamp_ex（1 字节）表示时间戳的扩展字节，当 24 位数值不够时，该字节最为最高位将时间戳扩展为 32 位数值</td></tr>
<tr><td>StreamID（3 字节）表示 stream id 总是 0</td></tr>
<tr><td>Tag Data</td><td>不同类型 Tag 的 data 部分结构各不相同，当 header 的结构是相同的</td></tr>
<tr><td colspan="3">Previous Tag size #1 即 Tag #1 的大小（11 + Datasize）</td></tr>
<tr><td colspan="3">Tag #2</td></tr>
<tr><td colspan="3">Previous Tag size #2</td></tr>
<tr><td colspan="3">… …</td></tr>
<tr><td colspan="3">Tag #N</td></tr>
<tr><td colspan="3">Previous Tag size #N</td></tr>
</table>

从FLV的封装格式结构可以看出，它的文件数据是一个一个的Tag连接起来的，中间间隔包含着Previous Tag Size。因此，flv_write_packet()函数的任务就是写入一个Tag和Previous Tag Size。下面简单记录一下Tag Data的格式。Tag Data根据Tag的Type不同而不同：可以分为音频Tag Data，视频Tag Data以及Script Tag Data。下面简述一下音频Tag Data和视频Tag Data。

Audio Tag Data

Audio Tag在官方标准中定义如下。

AUDIODATA

Field	Type	Comment
SoundFormat (see notes following table)	UB[4] 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 14 = MP3 8-Khz 15 = Device-specific sound	Format of SoundData Formats 7, 8, 14, and 15 are reserved for internal use AAC is supported in Flash Player 9,0,115,0 and higher. Speex is supported in Flash Player 10 and higher.
SoundRate	UB[2] 0 = 5.5-kHz 1 = 11-kHz 2 = 22-kHz 3 = 44-kHz	Sampling rate For AAC: always 3
SoundSize	UB[1] 0 = snd8Bit 1 = snd16Bit	Size of each sample. This parameter only pertains to uncompressed formats. Compressed formats always decode to 16 bits internally. 0 = snd8Bit 1 = snd16Bit
SoundType	UB[1] 0 = sndMono 1 = sndStereo	Mono or stereo sound For Nellymoser: always 0 For AAC: always 1
SoundData	UI8[size of sound data]	if SoundFormat == 10 AACAUDIODATA else Sound data—varies by format

Audio Tag开始的第1个字节包含了音频数据的参数信息，从第2个字节开始为音频流数据。

第1个字节的前4位的数值表示了音频数据格式：

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

14 = MP3 8-Khz

15 = Device-specific sound

第1个字节的第5-6位的数值表示采样率：0 = 5.5kHz，1 = 11KHz，2 = 22 kHz，3 = 44 kHz。

第1个字节的第7位表示采样精度：0 = 8bits，1 = 16bits。

第1个字节的第8位表示音频类型：0 = sndMono，1 = sndStereo。

其中，当音频编码为AAC的时候，第一个字节后面存储的是AACAUDIODATA，格式如下所示。

AACAUDIODATA

Field	Type	Comment
AACPacketType	UI8	0: AAC sequence header 1: AAC raw
Data	UI8[n]	if AACPacketType == 0 AudioSpecificConfig else if AACPacketType == 1 Raw AAC frame data

Video Tag Data

Video Tag在官方标准中的定义如下。

VIDEODATA

VIDEODATA		
Field	**Type**	**Comment**
FrameType	UB[4]	1: keyframe (for AVC, a seekable frame) 2: inter frame (for AVC, a non-seekable frame) 3: disposable inter frame (H.263 only) 4: generated keyframe (reserved for server use only) 5: video info/command frame
CodecID	UB[4]	1: JPEG (currently unused) 2: Sorenson H.263 3: Screen video 4: On2 VP6 5: On2 VP6 with alpha channel 6: Screen video version 2 7: AVC
VideoData	If CodecID == 2 H263VIDEOPACKET If CodecID == 3 SCREENVIDEOPACKET If CodecID == 4 VP6FLVVIDEOPACKET If CodecID == 5 VP6FLVALPHAVIDEOPACKET If CodecID == 6 SCREENV2VIDEOPACKET if CodecID == 7 AVCVIDEOPACKET	Video frame payload or UI8 (see note following table)

Video Tag也用开始的第1个字节包含视频数据的参数信息，从第2个字节为视频流数据。
第1个字节的前4位的数值表示帧类型（FrameType）：

1: keyframe (for AVC, a seekableframe)（关键帧）

2: inter frame (for AVC, a nonseekableframe)

3: disposable inter frame (H.263only)

4: generated keyframe (reservedfor server use only)

5: video info/command frame

第1个字节的后4位的数值表示视频编码ID（CodecID）：

1: JPEG (currently unused)

2: Sorenson H.263

3: Screen video

4: On2 VP6

5: On2 VP6 with alpha channel

6: Screen video version 2

7: AVC

其中，当音频编码为AVC（H.264）的时候，第一个字节后面存储的是AVCVIDEOPACKET，格式如下所示。

AVCVIDEOPACKET		
Field	**Type**	**Comment**
AVCPacketType	UI8	0: AVC sequence header 1: AVC NALU 2: AVC end of sequence (lower level NALU sequence ender is not required or supported)
CompositionTime	SI24	if AVCPacketType == 1 Composition time offset else 0
Data	UI8[n]	if AVCPacketType == 0 AVCDecoderConfigurationRecord else if AVCPacketType == 1 One or more NALUs (can be individual slices per FLV packets; that is, full frames are not strictly required) else if AVCPacketType == 2 Empty

flv_write_packet()

下面我们看一下FLV格式中write_packet()对应的实现函数flv_write_packet()的定义，位于libavformat\flvenc.c，如下所示。

[cpp]

```cpp
static int flv_write_packet(AVFormatContext *s, AVPacket *pkt)
{
    AVIOContext *pb      = s->pb;
    AVCodecContext *enc  = s->streams[pkt->stream_index]->codec;
```

```
    FLVContext *flv      = s->priv_data;
    FLVStreamContext *sc = s->streams[pkt->stream_index]->priv_data;
    unsigned ts;
    int size = pkt->size;
    uint8_t *data = NULL;
    int flags = -1, flags_size, ret;

    if (enc->codec_id == AV_CODEC_ID_VP6F || enc->codec_id == AV_CODEC_ID_VP6A ||
        enc->codec_id == AV_CODEC_ID_VP6  || enc->codec_id == AV_CODEC_ID_AAC)
        flags_size = 2;
    else if (enc->codec_id == AV_CODEC_ID_H264 || enc->codec_id == AV_CODEC_ID_MPEG4)
        flags_size = 5;
    else
        flags_size = 1;

    if (flv->delay == AV_NOPTS_VALUE)
        flv->delay = -pkt->dts;

    if (pkt->dts < -flv->delay) {
        av_log(s, AV_LOG_WARNING,
               "Packets are not in the proper order with respect to DTS\n");
        return AVERROR(EINVAL);
    }

    ts = pkt->dts + flv->delay; // add delay to force positive dts

    if (s->event_flags & AVSTREAM_EVENT_FLAG_METADATA_UPDATED) {
        write_metadata(s, ts);
        s->event_flags &= ~AVSTREAM_EVENT_FLAG_METADATA_UPDATED;
    }
    //Tag Header
    switch (enc->codec_type) {
    case AVMEDIA_TYPE_VIDEO:
        //Type
        avio_w8(pb, FLV_TAG_TYPE_VIDEO);

        flags = enc->codec_tag;
        if (flags == 0) {
            av_log(s, AV_LOG_ERROR,
                   "Video codec '%s' is not compatible with FLV\n",
                   avcodec_get_name(enc->codec_id));
            return AVERROR(EINVAL);
        }
        //Key Frame?
        flags |= pkt->flags & AV_PKT_FLAG_KEY ? FLV_FRAME_KEY : FLV_FRAME_INTER;
        break;
    case AVMEDIA_TYPE_AUDIO:

        flags = get_audio_flags(s, enc);

        av_assert0(size);
        //Type
        avio_w8(pb, FLV_TAG_TYPE_AUDIO);
        break;
    case AVMEDIA_TYPE_DATA:
        //Type
        avio_w8(pb, FLV_TAG_TYPE_META);
        break;
    default:
        return AVERROR(EINVAL);
    }

    if (enc->codec_id == AV_CODEC_ID_H264 || enc->codec_id == AV_CODEC_ID_MPEG4) {
        /* check if extradata looks like mp4 formated */
        if (enc->extradata_size > 0 && *(uint8_t*)enc->extradata != 1)
            if ((ret = ff_avc_parse_nal_units_buf(pkt->data, &data, &size)) < 0)
                return ret;
    } else if (enc->codec_id == AV_CODEC_ID_AAC && pkt->size > 2 &&
               (AV_RB16(pkt->data) & 0xfff0) == 0xfff0) {
        if (!s->streams[pkt->stream_index]->nb_frames) {
        av_log(s, AV_LOG_ERROR, "Malformed AAC bitstream detected: "
               "use the audio bitstream filter 'aac_adtstoasc' to fix it "
               "('-bsf:a aac_adtstoasc' option with ffmpeg)\n");
        return AVERROR_INVALIDDATA;
        }
        av_log(s, AV_LOG_WARNING, "aac bitstream error\n");
    }

    /* check Speex packet duration */
    if (enc->codec_id == AV_CODEC_ID_SPEEX && ts - sc->last_ts > 160)
        av_log(s, AV_LOG_WARNING, "Warning: Speex stream has more than "
                                  "8 frames per packet. Adobe Flash "
                                  "Player cannot handle this!\n");

    if (sc->last_ts < ts)
        sc->last_ts = ts;

    if (size + flags_size >= 1<<24) {
        av_log(s, AV_LOG_ERROR, "Too large packet with size %u >= %u\n",
               size + flags_size, 1<<24);
        return AVERROR(EINVAL);
```

```
    }
    //Tag Header - Datasize
    avio_wb24(pb, size + flags_size);
    //Tag Header - Timestamp
    avio_wb24(pb, ts & 0xFFFFFF);
    avio_w8(pb, (ts >> 24) & 0x7F); // timestamps are 32 bits _signed_
    //StreamID
    avio_wb24(pb, flv->reserved);

    if (enc->codec_type == AVMEDIA_TYPE_DATA) {
        int data_size;
        int64_t metadata_size_pos = avio_tell(pb);
        if (enc->codec_id == AV_CODEC_ID_TEXT) {
            // legacy FFmpeg magic?
            avio_w8(pb, AMF_DATA_TYPE_STRING);
            put_amf_string(pb, "onTextData");
            avio_w8(pb, AMF_DATA_TYPE_MIXEDARRAY);
            avio_wb32(pb, 2);
            put_amf_string(pb, "type");
            avio_w8(pb, AMF_DATA_TYPE_STRING);
            put_amf_string(pb, "Text");
            put_amf_string(pb, "text");
            avio_w8(pb, AMF_DATA_TYPE_STRING);
            put_amf_string(pb, pkt->data);
            put_amf_string(pb, "");
            avio_w8(pb, AMF_END_OF_OBJECT);
        } else {
            // just pass the metadata through
            avio_write(pb, data ? data : pkt->data, size);
        }
        /* write total size of tag */
        data_size = avio_tell(pb) - metadata_size_pos;
        avio_seek(pb, metadata_size_pos - 10, SEEK_SET);
        avio_wb24(pb, data_size);
        avio_seek(pb, data_size + 10 - 3, SEEK_CUR);
        avio_wb32(pb, data_size + 11);
    } else {
        av_assert1(flags>=0);
        //First Byte of Tag Data
        avio_w8(pb,flags);
        if (enc->codec_id == AV_CODEC_ID_VP6)
            avio_w8(pb,0);
        if (enc->codec_id == AV_CODEC_ID_VP6F || enc->codec_id == AV_CODEC_ID_VP6A) {
            if (enc->extradata_size)
                avio_w8(pb, enc->extradata[0]);
            else
                avio_w8(pb, ((FFALIGN(enc->width,  16) - enc->width) << 4) |
                             (FFALIGN(enc->height, 16) - enc->height));
        } else if (enc->codec_id == AV_CODEC_ID_AAC)
            avio_w8(pb, 1); // AAC raw
        else if (enc->codec_id == AV_CODEC_ID_H264 || enc->codec_id == AV_CODEC_ID_MPEG4) {
            //AVCVIDEOPACKET-AVCPacketType
            avio_w8(pb, 1); // AVC NALU
            //AVCVIDEOPACKET-CompositionTime
            avio_wb24(pb, pkt->pts - pkt->dts);
        }
        //Data
        avio_write(pb, data ? data : pkt->data, size);

        avio_wb32(pb, size + flags_size + 11); // previous tag size
        flv->duration = FFMAX(flv->duration,
                              pkt->pts + flv->delay + pkt->duration);
    }

    av_free(data);

    return pb->error;
}
```

我们通过源代码简单梳理一下flv_write_packet()在写入H.264/AAC时候的流程：
（1）写入Tag Header的Type，如果是视频，代码如下：

[cpp]

```
avio_w8(pb, FLV_TAG_TYPE_VIDEO);
```

如果是音频，代码如下：

[cpp]

```
avio_w8(pb, FLV_TAG_TYPE_AUDIO);
```

（2）写入Tag Header的Datasize，Timestamp和StreamID（至此完成Tag Header）：

[cpp]

```cpp
//Tag Header - Datasize
avio_wb24(pb, size + flags_size);
//Tag Header - Timestamp
avio_wb24(pb, ts & 0xFFFFFF);
avio_w8(pb, (ts >> 24) & 0x7F); // timestamps are 32 bits _signed_
//StreamID
avio_wb24(pb, flv->reserved);
```

（3）写入Tag Data的第一字节（其中flag已经在前面的代码中设置完毕）：

[cpp]

```cpp
//First Byte of Tag Data
avio_w8(pb,flags);
```

（4）如果编码格式VP6作相应的处理（不研究）；编码格式为AAC，写入AACAUDIODATA；编码格式为H.264，写入AVCVIDEOPACKET：

[cpp]

```cpp
if (enc->codec_id == AV_CODEC_ID_VP6F || enc->codec_id == AV_CODEC_ID_VP6A) {
    if (enc->extradata_size)
        avio_w8(pb, enc->extradata[0]);
    else
        avio_w8(pb, ((FFALIGN(enc->width,  16) - enc->width) << 4) |
                     (FFALIGN(enc->height, 16) - enc->height));
} else if (enc->codec_id == AV_CODEC_ID_AAC)
    avio_w8(pb, 1); // AAC raw
else if (enc->codec_id == AV_CODEC_ID_H264 || enc->codec_id == AV_CODEC_ID_MPEG4) {
    //AVCVIDEOPACKET-AVCPacketType
    avio_w8(pb, 1); // AVC NALU
    //AVCVIDEOPACKET-CompositionTime
    avio_wb24(pb, pkt->pts - pkt->dts);
}
```

（5）写入数据：

[cpp]

```cpp
//Data
avio_write(pb, data ? data : pkt->data, size);
```

（6）

写入previous tag size：

[cpp]

```cpp
avio_wb32(pb, size + flags_size + 11); // previous tag size
```

至此，flv_write_packet()就完成了一个Tag的写入。

雷霄骅
leixiaohua1020@126.com
http://blog.csdn.net/leixiaohua1020

文章标签： ffmpeg AVPacket FLV 输出 源代码

个人分类： FFMPEG

所属专栏： FFmpeg