原 ffmpeg 源代码简单分析 ： av_read_frame()

2013年10月13日 15:58:49　阅读数：61095

===

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

===

ffmpeg中的av_read_frame()的作用是读取码流中的音频若干帧或者视频一帧。例如，解码视频的时候，每解码一个视频帧，需要先调用 av_read_frame()获得一帧视频的压缩数据，然后才能对该数据进行解码（例如H.264中一帧压缩数据通常对应一个NAL）。

对该函数源代码的分析是很久之前做的了，现在翻出来，用博客记录一下。

上代码之前，先参考了其他人对av_read_frame()的解释，在此做一个参考：

> 通过av_read_packet(***)，读取一个包，需要说明的是此函数必须是包含整数帧的，不存在半帧的情况，以ts流为例，是读取一个完整的PES包（一个完整pes包包含若干视频或音频es包），读取完毕后，通过av_parser_parse2(***)分析出视频一帧（或音频若干帧），返回，下次进入循环的时候，如果上次的数据没有完全取完，则st = s->cur_st;不会是NULL，即再此进入av_parser_parse2(***)流程，而不是下面的av_read_packet（**）流程，这样就保证了，如果读取一次包含了N帧视频数据（以视频为例），则调用av_read_frame（***）N次都不会去读数据，而是返回第一次读取的数据，直到全部解析完毕。

av_read_frame()的声明位于libavformat\avformat.h，如下所示。

[cpp]

```cpp
/**
 * Return the next frame of a stream.
 * This function returns what is stored in the file, and does not validate
 * that what is there are valid frames for the decoder. It will split what is
 * stored in the file into frames and return one for each call. It will not
 * omit invalid data between valid frames so as to give the decoder the maximum
 * information possible for decoding.
 *
 * If pkt->buf is NULL, then the packet is valid until the next
 * av_read_frame() or until avformat_close_input(). Otherwise the packet
 * is valid indefinitely. In both cases the packet must be freed with
 * av_free_packet when it is no longer needed. For video, the packet contains
 * exactly one frame. For audio, it contains an integer number of frames if each
 * frame has a known fixed size (e.g. PCM or ADPCM data). If the audio frames
 * have a variable size (e.g. MPEG audio), then it contains one frame.
 *
 * pkt->pts, pkt->dts and pkt->duration are always set to correct
 * values in AVStream.time_base units (and guessed if the format cannot
 * provide them). pkt->pts can be AV_NOPTS_VALUE if the video format
 * has B-frames, so it is better to rely on pkt->dts if you do not
 * decompress the payload.
 *
 * @return 0 if OK, < 0 on error or end of file
 */
int av_read_frame(AVFormatContext *s, AVPacket *pkt);
```

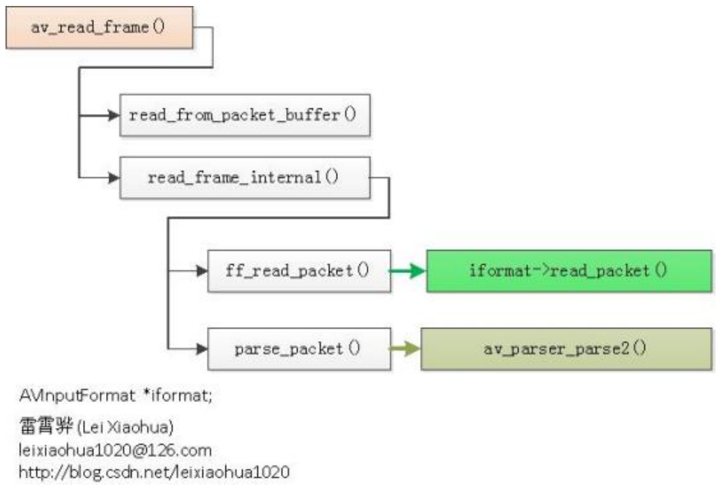
av_read_frame()使用方法在注释中写得很详细，用中文简单描述一下它的两个参数：

> s：输入的AVFormatContext
>
> pkt：输出的AVPacket

如果返回0则说明读取正常。

函数调用结构图

函数调用结构图如下所示。



av_read_frame()

av_read_frame()的定义位于libavformat\utils.c，如下所示：

[cpp]

```cpp
//获取一个AVPacket
/*
 * av_read_frame - 新版本的ffmpeg用的是av_read_frame，而老版本的是av_read_packet
 * 。区别是av_read_packet读出的是包，它可能是半帧或多帧，不保证帧的完整性。av_read_frame对
 * av_read_packet进行了封装，使读出的数据总是完整的帧
 */
int av_read_frame(AVFormatContext *s, AVPacket *pkt)
{
    const int genpts = s->flags & AVFMT_FLAG_GENPTS;
    int          eof = 0;

    if (!genpts)
        /**
         * This buffer is only needed when packets were already buffered but
         * not decoded, for example to get the codec parameters in MPEG
         * streams.
         * 一般情况下会调用read_frame_internal(s, pkt)
         * 直接返回
         */
        return s->packet_buffer ? read_from_packet_buffer(s, pkt) :
                                  read_frame_internal(s, pkt);

    for (;;) {
        int ret;
        AVPacketList *pktl = s->packet_buffer;

        if (pktl) {
            AVPacket *next_pkt = &pktl->pkt;

            if (next_pkt->dts != AV_NOPTS_VALUE) {
                int wrap_bits = s->streams[next_pkt->stream_index]->pts_wrap_bits;
                while (pktl && next_pkt->pts == AV_NOPTS_VALUE) {
                    if (pktl->pkt.stream_index == next_pkt->stream_index &&
                        (av_compare_mod(next_pkt->dts, pktl->pkt.dts, 2LL << (wrap_bits - 1)) < 0) &&
                         av_compare_mod(pktl->pkt.pts, pktl->pkt.dts, 2LL << (wrap_bits - 1))) { //not b frame
                        next_pkt->pts = pktl->pkt.dts;
                    }
                    pktl = pktl->next;
                }
                pktl = s->packet_buffer;
            }

            /* read packet from packet buffer, if there is data */
            if (!(next_pkt->pts == AV_NOPTS_VALUE &&
                  next_pkt->dts != AV_NOPTS_VALUE && !eof))
                return read_from_packet_buffer(s, pkt);
        }

        ret = read_frame_internal(s, pkt);
        if (ret < 0) {
            if (pktl && ret != AVERROR(EAGAIN)) {
                eof = 1;
                continue;
            } else
                return ret;
        }

        if (av_dup_packet(add_to_pktbuf(&s->packet_buffer, pkt,
                          &s->packet_buffer_end)) < 0)
            return AVERROR(ENOMEM);
    }
}
```

可以从源代码中看出，av_read_frame()调用了read_frame_internal()。

read_frame_internal()

read_frame_internal()代码如下所示：

[cpp]

```cpp
//av_read_frame对他进行了封装
static int read_frame_internal(AVFormatContext *s, AVPacket *pkt)
{
    int ret = 0, i, got_packet = 0;
    AVDictionary *metadata = NULL;
    //初始化
    av_init_packet(pkt);

    while (!got_packet && !s->parse_queue) {
        AVStream *st;
        AVPacket cur_pkt;

        /* read next packet */
        ret = ff_read_packet(s, &cur_pkt);
        if (ret < 0) {
            if (ret == AVERROR(EAGAIN))
                return ret;
            /* flush the parsers */
            for (i = 0; i < s->nb_streams; i++) {
                st = s->streams[i];
                //需要解析
                if (st->parser && st->need_parsing)
                    parse_packet(s, NULL, st->index);
            }
            /* all remaining packets are now in parse_queue =>
             * really terminate parsing */
            break;
        }
        ret = 0;
        st  = s->streams[cur_pkt.stream_index];

        if (cur_pkt.pts != AV_NOPTS_VALUE &&
            cur_pkt.dts != AV_NOPTS_VALUE &&
            cur_pkt.pts < cur_pkt.dts) {
            av_log(s, AV_LOG_WARNING,
                   "Invalid timestamps stream=%d, pts=%s, dts=%s, size=%d\n",
                   cur_pkt.stream_index,
                   av_ts2str(cur_pkt.pts),
                   av_ts2str(cur_pkt.dts),
                   cur_pkt.size);
        }
        if (s->debug & FF_FDEBUG_TS)
            av_log(s, AV_LOG_DEBUG,
                   "ff_read_packet stream=%d, pts=%s, dts=%s, size=%d, duration=%d, flags=%d\n",
                   cur_pkt.stream_index,
                   av_ts2str(cur_pkt.pts),
                   av_ts2str(cur_pkt.dts),
                   cur_pkt.size, cur_pkt.duration, cur_pkt.flags);

        if (st->need_parsing && !st->parser && !(s->flags & AVFMT_FLAG_NOPARSE)) {
            st->parser = av_parser_init(st->codec->codec_id);
            if (!st->parser) {
                av_log(s, AV_LOG_VERBOSE, "parser not found for codec "
                       "%s, packets or times may be invalid.\n",
                       avcodec_get_name(st->codec->codec_id));
                /* no parser available: just output the raw packets */
                st->need_parsing = AVSTREAM_PARSE_NONE;
            } else if (st->need_parsing == AVSTREAM_PARSE_HEADERS)
                st->parser->flags |= PARSER_FLAG_COMPLETE_FRAMES;
            else if (st->need_parsing == AVSTREAM_PARSE_FULL_ONCE)
                st->parser->flags |= PARSER_FLAG_ONCE;
            else if (st->need_parsing == AVSTREAM_PARSE_FULL_RAW)
                st->parser->flags |= PARSER_FLAG_USE_CODEC_TS;
        }
        if (!st->need_parsing || !st->parser) {
            /* no parsing needed: we just output the packet as is */
            *pkt = cur_pkt;
            compute_pkt_fields(s, st, NULL, pkt);
            if ((s->iformat->flags & AVFMT_GENERIC_INDEX) &&
                (pkt->flags & AV_PKT_FLAG_KEY) && pkt->dts != AV_NOPTS_VALUE) {
                ff_reduce_index(s, st->index);
                av_add_index_entry(st, pkt->pos, pkt->dts,
                                   0, 0, AVINDEX_KEYFRAME);
            }
            got_packet = 1;
        } else if (st->discard < AVDISCARD_ALL) {
            if ((ret = parse_packet(s, &cur_pkt, cur_pkt.stream_index)) < 0)
                return ret;
        } else {
            /* free packet */
            av_free_packet(&cur_pkt);
        }
        if (pkt->flags & AV_PKT_FLAG_KEY)
```

```c
            st->skip_to_keyframe = 0;
        if (st->skip_to_keyframe) {
            av_free_packet(&cur_pkt);
            if (got_packet) {
                *pkt = cur_pkt;
            }
            got_packet = 0;
        }
    }

    if (!got_packet && s->parse_queue)
        ret = read_from_packet_buffer(&s->parse_queue, &s->parse_queue_end, pkt);

    if (ret >= 0) {
        AVStream *st = s->streams[pkt->stream_index];
        int discard_padding = 0;
        if (st->first_discard_sample && pkt->pts != AV_NOPTS_VALUE) {
            int64_t pts = pkt->pts - (is_relative(pkt->pts) ? RELATIVE_TS_BASE : 0);
            int64_t sample = ts_to_samples(st, pts);
            int duration = ts_to_samples(st, pkt->duration);
            int64_t end_sample = sample + duration;
            if (duration > 0 && end_sample >= st->first_discard_sample &&
                sample < st->last_discard_sample)
                discard_padding = FFMIN(end_sample - st->first_discard_sample, duration);
        }
        if (st->skip_samples || discard_padding) {
            uint8_t *p = av_packet_new_side_data(pkt, AV_PKT_DATA_SKIP_SAMPLES, 10);
            if (p) {
                AV_WL32(p, st->skip_samples);
                AV_WL32(p + 4, discard_padding);
                av_log(s, AV_LOG_DEBUG, "demuxer injecting skip %d\n", st->skip_samples);
            }
            st->skip_samples = 0;
        }

        if (st->inject_global_side_data) {
            for (i = 0; i < st->nb_side_data; i++) {
                AVPacketSideData *src_sd = &st->side_data[i];
                uint8_t *dst_data;

                if (av_packet_get_side_data(pkt, src_sd->type, NULL))
                    continue;

                dst_data = av_packet_new_side_data(pkt, src_sd->type, src_sd->size);
                if (!dst_data) {
                    av_log(s, AV_LOG_WARNING, "Could not inject global side data\n");
                    continue;
                }

                memcpy(dst_data, src_sd->data, src_sd->size);
            }
            st->inject_global_side_data = 0;
        }

        if (!(s->flags & AVFMT_FLAG_KEEP_SIDE_DATA))
            av_packet_merge_side_data(pkt);
    }

    av_opt_get_dict_val(s, "metadata", AV_OPT_SEARCH_CHILDREN, &metadata);
    if (metadata) {
        s->event_flags |= AVFMT_EVENT_FLAG_METADATA_UPDATED;
        av_dict_copy(&s->metadata, metadata, 0);
        av_dict_free(&metadata);
        av_opt_set_dict_val(s, "metadata", NULL, AV_OPT_SEARCH_CHILDREN);
    }

    if (s->debug & FF_FDEBUG_TS)
        av_log(s, AV_LOG_DEBUG,
               "read_frame_internal stream=%d, pts=%s, dts=%s, "
               "size=%d, duration=%d, flags=%d\n",
               pkt->stream_index,
               av_ts2str(pkt->pts),
               av_ts2str(pkt->dts),
               pkt->size, pkt->duration, pkt->flags);

    return ret;
}
```

read_frame_internal()代码比较长，这里只简单看一下它前面的部分。它前面部分有2步是十分关键的：
（1）调用了ff_read_packet()从相应的AVInputFormat读取数据。

（2）如果媒体频流需要使用AVCodecParser，则调用parse_packet()解析相应的AVPacket。

下面我们分成分别看一下ff_read_packet()和parse_packet()的源代码。

ff_read_packet()

ff_read_packet()的代码比较长，如下所示。

[cpp]

```cpp
int ff_read_packet(AVFormatContext *s, AVPacket *pkt)
{
    int ret, i, err;
    AVStream *st;

    for (;;) {
        AVPacketList *pktl = s->raw_packet_buffer;

        if (pktl) {
            *pkt = pktl->pkt;
            st   = s->streams[pkt->stream_index];
            if (s->raw_packet_buffer_remaining_size <= 0)
                if ((err = probe_codec(s, st, NULL)) < 0)
                    return err;
            if (st->request_probe <= 0) {
                s->raw_packet_buffer                 = pktl->next;
                s->raw_packet_buffer_remaining_size += pkt->size;
                av_free(pktl);
                return 0;
            }
        }

        pkt->data = NULL;
        pkt->size = 0;
        av_init_packet(pkt);
        //关键：读取Packet
        ret = s->iformat->read_packet(s, pkt);
        if (ret < 0) {
            if (!pktl || ret == AVERROR(EAGAIN))
                return ret;
            for (i = 0; i < s->nb_streams; i++) {
                st = s->streams[i];
                if (st->probe_packets)
                    if ((err = probe_codec(s, st, NULL)) < 0)
                        return err;
                av_assert0(st->request_probe <= 0);
            }
            continue;
        }

        if ((s->flags & AVFMT_FLAG_DISCARD_CORRUPT) &&
            (pkt->flags & AV_PKT_FLAG_CORRUPT)) {
            av_log(s, AV_LOG_WARNING,
                   "Dropped corrupted packet (stream = %d)\n",
                   pkt->stream_index);
            av_free_packet(pkt);
            continue;
        }

        if (pkt->stream_index >= (unsigned)s->nb_streams) {
            av_log(s, AV_LOG_ERROR, "Invalid stream index %d\n", pkt->stream_index);
            continue;
        }

        st = s->streams[pkt->stream_index];

        if (update_wrap_reference(s, st, pkt->stream_index, pkt) && st->pts_wrap_behavior == AV_PTS_WRAP_SUB_OFFSET) {
            // correct first time stamps to negative values
            if (!is_relative(st->first_dts))
                st->first_dts = wrap_timestamp(st, st->first_dts);
            if (!is_relative(st->start_time))
                st->start_time = wrap_timestamp(st, st->start_time);
            if (!is_relative(st->cur_dts))
                st->cur_dts = wrap_timestamp(st, st->cur_dts);
        }

        pkt->dts = wrap_timestamp(st, pkt->dts);
        pkt->pts = wrap_timestamp(st, pkt->pts);

        force_codec_ids(s, st);

        /* TODO: audio: time filter; video: frame reordering (pts != dts) */
        if (s->use_wallclock_as_timestamps)
            pkt->dts = pkt->pts = av_rescale_q(av_gettime(), AV_TIME_BASE_Q, st->time_base);

        if (!pktl && st->request_probe <= 0)
            return ret;

        add_to_pktbuf(&s->raw_packet_buffer, pkt, &s->raw_packet_buffer_end);
        s->raw_packet_buffer_remaining_size -= pkt->size;

        if ((err = probe_codec(s, st, pkt)) < 0)
            return err;
    }
}
```

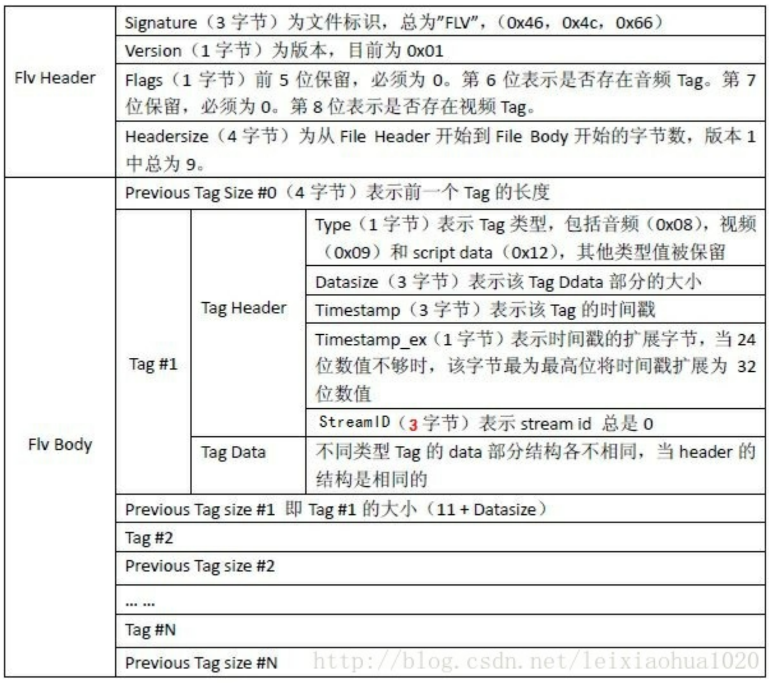
ff_read_packet()中最关键的地方就是调用了AVInputFormat的read_packet()方法。AVInputFormat的read_packet()是一个函数指针，指向当前的AVInputFormat的读取

数据的函数。在这里我们以FLV封装格式对应的AVInputFormat为例，看看read_packet()的实现函数是什么样子的。

FLV封装格式对应的AVInputFormat的定义位于libavformat\flvdec.c，如下所示。

[cpp]
```cpp
AVInputFormat ff_flv_demuxer = {
    .name           = "flv",
    .long_name      = NULL_IF_CONFIG_SMALL("FLV (Flash Video)"),
    .priv_data_size = sizeof(FLVContext),
    .read_probe     = flv_probe,
    .read_header    = flv_read_header,
    .read_packet    = flv_read_packet,
    .read_seek      = flv_read_seek,
    .read_close     = flv_read_close,
    .extensions     = "flv",
    .priv_class     = &flv_class,
};
```

从ff_flv_demuxer的定义可以看出，read_packet()对应的是flv_read_packet()函数。在看flv_read_packet()函数之前，我们先回顾一下FLV封装格式的结构，如下图所示。

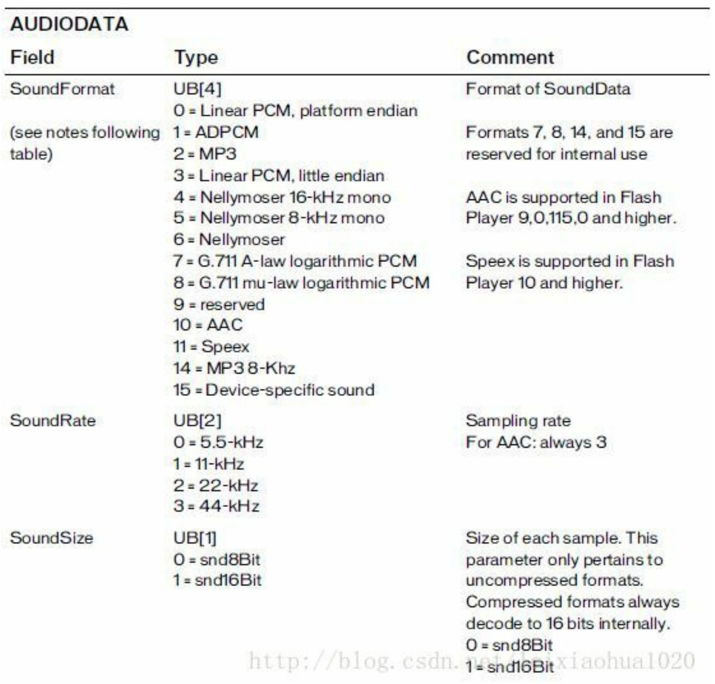


Flv Header	Signature（3 字节）为文件标识，总为"FLV"，（0x46，0x4c，0x66）		
	Version（1 字节）为版本，目前为 0x01		
	Flags（1 字节）前 5 位保留，必须为 0。第 6 位表示是否存在音频 Tag。第 7 位保留，必须为 0。第 8 位表示是否存在视频 Tag。		
	Headersize（4 字节）为从 File Header 开始到 File Body 开始的字节数，版本 1 中总为 9。		
Flv Body	Previous Tag Size #0（4 字节）表示前一个 Tag 的长度		
	Tag #1	Tag Header	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
			Datasize（3 字节）表示该 Tag Ddata 部分的大小
			Timestamp（3 字节）表示该 Tag 的时间戳
			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
			StreamID（3 字节）表示 stream id 总是 0
		Tag Data	不同类型 Tag 的 data 部分结构各不相同，当 header 的结构是相同的
	Previous Tag size #1 即 Tag #1 的大小（11 + Datasize）		
	Tag #2		
	Previous Tag size #2		
	… …		
	Tag #N		
	Previous Tag size #N		

从图中可以看出，FLV文件体部分是由一个一个的Tag连接起来的（中间间隔着Previous Tag Size）。每个Tag包含了Tag Header和Tag Data两个部分。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

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

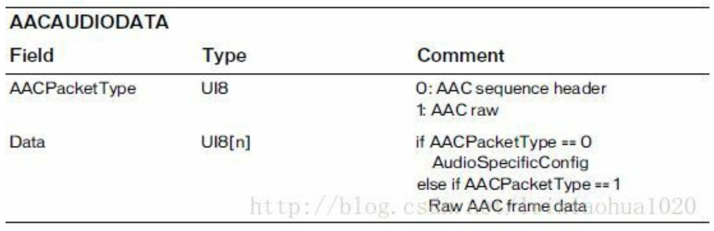
9 = reserved

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。



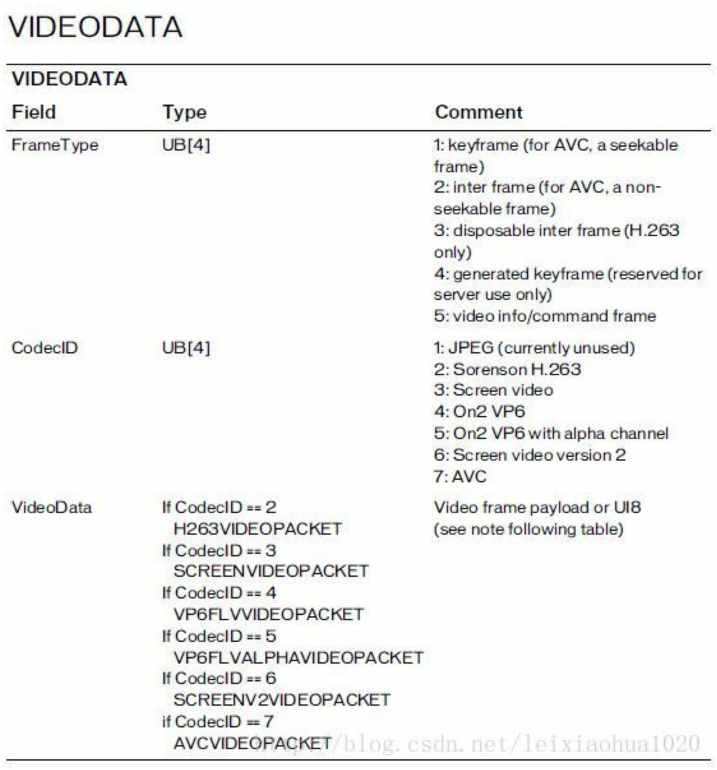
AACAUDIODATA

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

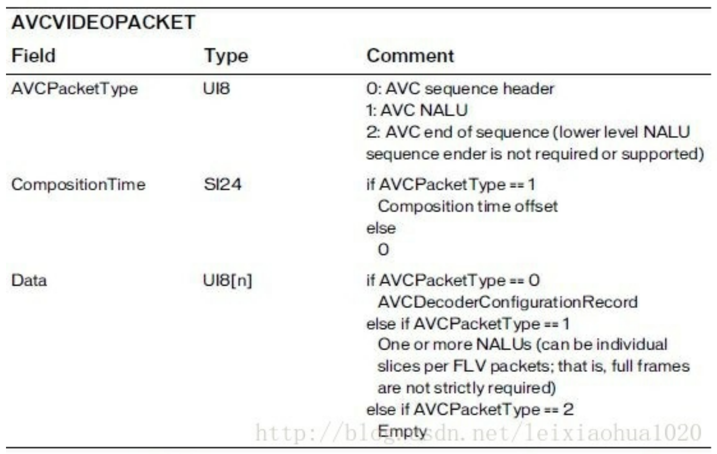
4: On2 VP6

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

了解了FLV的基本格式之后，就可以看一下FLV解析Tag的函数flv_read_packet()了。

flv_read_packet()

flv_read_packet()的定义位于libavformat\flvdec.c，如下所示。

[cpp]

```cpp
static int flv_read_packet(AVFormatContext *s, AVPacket *pkt)
{
```

```c
    FLVContext *flv = s->priv_data;
    int ret, i, type, size, flags;
    int stream_type=-1;
    int64_t next, pos, meta_pos;
    int64_t dts, pts = AV_NOPTS_VALUE;
    int av_uninit(channels);
    int av_uninit(sample_rate);
    AVStream *st    = NULL;

    /* pkt size is repeated at end. skip it */
    for (;; avio_skip(s->pb, 4)) {
        pos  = avio_tell(s->pb);
        //解析Tag Header==========
        //Tag类型
        type = (avio_r8(s->pb) & 0x1F);
        //Datasize数据大小
        size = avio_rb24(s->pb);
        //Timstamp时间戳
        dts  = avio_rb24(s->pb);
        dts |= avio_r8(s->pb) << 24;
        av_dlog(s, "type:%d, size:%d, dts:%"PRId64" pos:%"PRId64"\n", type, size, dts, avio_tell(s->pb));
        if (avio_feof(s->pb))
            return AVERROR_EOF;
        //StreamID
        avio_skip(s->pb, 3); /* stream id, always 0 */
        flags = 0;
        //========================
        if (flv->validate_next < flv->validate_count) {
            int64_t validate_pos = flv->validate_index[flv->validate_next].pos;
            if (pos == validate_pos) {
                if (FFABS(dts - flv->validate_index[flv->validate_next].dts) <=
                    VALIDATE_INDEX_TS_THRESH) {
                    flv->validate_next++;
                } else {
                    clear_index_entries(s, validate_pos);
                    flv->validate_count = 0;
                }
            } else if (pos > validate_pos) {
                clear_index_entries(s, validate_pos);
                flv->validate_count = 0;
            }
        }

        if (size == 0)
            continue;

        next = size + avio_tell(s->pb);

        if (type == FLV_TAG_TYPE_AUDIO) {
            //Type是音频
            stream_type = FLV_STREAM_TYPE_AUDIO;
            //Tag Data的第一个字节
            flags    = avio_r8(s->pb);
            size--;
        } else if (type == FLV_TAG_TYPE_VIDEO) {
            //Type是音频
            stream_type = FLV_STREAM_TYPE_VIDEO;
            //Tag Data的第一个字节
            flags    = avio_r8(s->pb);
            size--;
            if ((flags & FLV_VIDEO_FRAMETYPE_MASK) == FLV_FRAME_VIDEO_INFO_CMD)
                goto skip;
        } else if (type == FLV_TAG_TYPE_META) {
            stream_type=FLV_STREAM_TYPE_DATA;
            if (size > 13 + 1 + 4 && dts == 0) { // Header-type metadata stuff
                meta_pos = avio_tell(s->pb);
                if (flv_read_metabody(s, next) <= 0) {
                    goto skip;
                }
                avio_seek(s->pb, meta_pos, SEEK_SET);
            }
        } else {
            av_log(s, AV_LOG_DEBUG,
                   "Skipping flv packet: type %d, size %d, flags %d.\n",
                   type, size, flags);
skip:
            avio_seek(s->pb, next, SEEK_SET);
            continue;
        }

        /* skip empty data packets */
        if (!size)
            continue;

        /* now find stream */
        for (i = 0; i < s->nb_streams; i++) {
            st = s->streams[i];
            if (stream_type == FLV_STREAM_TYPE_AUDIO) {
                if (st->codec->codec_type == AVMEDIA_TYPE_AUDIO &&
                    (s->audio_codec_id || flv_same_audio_codec(st->codec, flags)))
                    break;
            } else if (stream_type == FLV_STREAM_TYPE_VIDEO) {
            } else if (stream_type == FLV_STREAM_TYPE_VIDEO) {
                if (st->codec->codec_type == AVMEDIA_TYPE_VIDEO &&
                    (s->video_codec_id || flv_same_video_codec(st->codec, flags)))
                    break;
            } else if (stream_type == FLV_STREAM_TYPE_DATA) {
                if (st->codec->codec_type == AVMEDIA_TYPE_DATA)
                    break;
            }
        }
        if (i == s->nb_streams) {
            static const enum AVMediaType stream_types[] = {AVMEDIA_TYPE_VIDEO, AVMEDIA_TYPE_AUDIO, AVMEDIA_TYPE_DATA};
            av_log(s, AV_LOG_WARNING, "Stream discovered after head already parsed\n");
            st = create_stream(s, stream_types[stream_type]);
            if (!st)
                return AVERROR(ENOMEM);

        }
        av_dlog(s, "%d %X %d \n", stream_type, flags, st->discard);

        if ((flags & FLV_VIDEO_FRAMETYPE_MASK) == FLV_FRAME_KEY ||
            stream_type == FLV_STREAM_TYPE_AUDIO)
            av_add_index_entry(st, pos, dts, size, 0, AVINDEX_KEYFRAME);

        if (  (st->discard >= AVDISCARD_NONKEY && !
((flags & FLV_VIDEO_FRAMETYPE_MASK) == FLV_FRAME_KEY || (stream_type == FLV_STREAM_TYPE_AUDIO)))
            ||(st-
>discard >= AVDISCARD_BIDIR  &&  ((flags & FLV_VIDEO_FRAMETYPE_MASK) == FLV_FRAME_DISP_INTER && (stream_type == FLV_STREAM_TYPE_VIDEO

            || st->discard >= AVDISCARD_ALL
        ) {
            avio_seek(s->pb, next, SEEK_SET);
            continue;
        }
        break;
    }

    // if not streamed and no duration from metadata then seek to end to find
    // the duration from the timestamps
    if (s->pb->seekable && (!s->duration || s->duration == AV_NOPTS_VALUE) && !flv->searched_for_end) {
        int size;
        const int64_t pos   = avio_tell(s->pb);
        // Read the last 4 bytes of the file, this should be the size of the
        // previous FLV tag. Use the timestamp of its payload as duration.
        int64_t fsize       = avio_size(s->pb);
retry_duration:
        avio_seek(s->pb, fsize - 4, SEEK_SET);
        size = avio_rb32(s->pb);
        // Seek to the start of the last FLV tag at position (fsize - 4 - size)
        // but skip the byte indicating the type.
        avio_seek(s->pb, fsize - 3 - size, SEEK_SET);
        if (size == avio_rb24(s->pb) + 11) {
            uint32_t ts = avio_rb24(s->pb);
            ts         |= avio_r8(s->pb) << 24;
            if (ts)
                s->duration = ts * (int64_t)AV_TIME_BASE / 1000;
            else if (fsize >= 8 && fsize - 8 >= size) {
                fsize -= size+4;
                goto retry_duration;
            }
        }

        avio_seek(s->pb, pos, SEEK_SET);
        flv->searched_for_end = 1;
    }

    if (stream_type == FLV_STREAM_TYPE_AUDIO) {
        int bits_per_coded_sample;
        channels = (flags & FLV_AUDIO_CHANNEL_MASK) == FLV_STEREO ? 2 : 1;
        sample_rate = 44100 << ((flags & FLV_AUDIO_SAMPLERATE_MASK) >>
                                FLV_AUDIO_SAMPLERATE_OFFSET) >> 3;
        bits_per_coded_sample = (flags & FLV_AUDIO_SAMPLESIZE_MASK) ? 16 : 8;
        if (!st->codec->channels || !st->codec->sample_rate ||
            !st->codec->bits_per_coded_sample) {
            st->codec->channels              = channels;
            st->codec->channel_layout        = channels == 1
                                               ? AV_CH_LAYOUT_MONO
                                               : AV_CH_LAYOUT_STEREO;
            st->codec->sample_rate           = sample_rate;
            st->codec->bits_per_coded_sample = bits_per_coded_sample;
        }
        if (!st->codec->codec_id) {
            flv_set_audio_codec(s, st, st->codec,
                                flags & FLV_AUDIO_CODECID_MASK);
            flv->last_sample_rate =
            sample_rate           = st->codec->sample_rate;
            flv->last_channels    =
            channels              = st->codec->channels;
        } else {
            AVCodecContext ctx = {0};
            ctx.sample_rate = sample_rate;
            ctx.bits_per_coded_sample = bits_per_coded_sample;
            flv_set_audio_codec(s, st, &ctx, flags & FLV_AUDIO_CODECID_MASK);
            sample_rate = ctx.sample_rate;
        }
    } else if (stream_type == FLV_STREAM_TYPE_VIDEO) {
        size -= flv_set_video_codec(s, st, flags & FLV_VIDEO_CODECID_MASK, 1);
    }
    //几种特殊的格式
    if (st->codec->codec_id == AV_CODEC_ID_AAC ||
        st->codec->codec_id == AV_CODEC_ID_H264 ||
        st->codec->codec_id == AV_CODEC_ID_MPEG4) {
        //对应AACPacketType或者AVCPacketType
        int type = avio_r8(s->pb);
        size--;
        //H.264
        if (st->codec->codec_id == AV_CODEC_ID_H264 || st->codec->codec_id == AV_CODEC_ID_MPEG4) {
            // sign extension
            //对应CompositionTime
            int32_t cts = (avio_rb24(s->pb) + 0xff800000) ^ 0xff800000;
            //计算PTS
            pts = dts + cts;
            if (cts < 0) { // dts might be wrong
                if (!flv->wrong_dts)
                    av_log(s, AV_LOG_WARNING,
                        "Negative cts, previous timestamps might be wrong.\n");
                flv->wrong_dts = 1;
            } else if (FFABS(dts - pts) > 1000*60*15) {
                av_log(s, AV_LOG_WARNING,
                       "invalid timestamps %"PRId64" %"PRId64"\n", dts, pts);
                dts = pts = AV_NOPTS_VALUE;
            }
        }
        //如果编码器是AAC或者H.264
        if (type == 0 && (!st->codec->extradata || st->codec->codec_id == AV_CODEC_ID_AAC ||
            st->codec->codec_id == AV_CODEC_ID_H264)) {
            AVDictionaryEntry *t;

            if (st->codec->extradata) {
                if ((ret = flv_queue_extradata(flv, s->pb, stream_type, size)) < 0)
                    return ret;
                ret = AVERROR(EAGAIN);
                goto leave;
            }
            if ((ret = flv_get_extradata(s, st, size)) < 0)
                return ret;

            /* Workaround for buggy Omnia A/XE encoder */
            t = av_dict_get(s->metadata, "Encoder", NULL, 0);
            if (st->codec->codec_id == AV_CODEC_ID_AAC && t && !strcmp(t->value, "Omnia A/XE"))
                st->codec->extradata_size = 2;
            //AAC
            if (st->codec->codec_id == AV_CODEC_ID_AAC && 0) {
                MPEG4AudioConfig cfg;

                if (avpriv_mpeg4audio_get_config(&cfg, st->codec->extradata,
                                             st->codec->extradata_size * 8, 1) >= 0) {
                st->codec->channels       = cfg.channels;
                st->codec->channel_layout = 0;
                if (cfg.ext_sample_rate)
                    st->codec->sample_rate = cfg.ext_sample_rate;
                else
                    st->codec->sample_rate = cfg.sample_rate;
                av_dlog(s, "mp4a config channels %d sample rate %d\n",
                        st->codec->channels, st->codec->sample_rate);
                }
            }

            ret = AVERROR(EAGAIN);
            goto leave;
        }
    }

    /* skip empty data packets */
    if (!size) {
        ret = AVERROR(EAGAIN);
        goto leave;
    }

    ret = av_get_packet(s->pb, pkt, size);
    if (ret < 0)
        return ret;
    //设置PTS、DTS等等
    pkt->dts          = dts;
    pkt->pts          = pts == AV_NOPTS_VALUE ? dts : pts;
    pkt->stream_index = st->index;
    if (flv->new_extradata[stream_type]) {
        uint8_t *side = av_packet_new_side_data(pkt, AV_PKT_DATA_NEW_EXTRADATA,
                                                flv->new_extradata_size[stream_type]);
        if (side) {
            memcpy(side, flv->new_extradata[stream_type],
                   flv->new_extradata_size[stream_type]);
            av_freep(&flv->new_extradata[stream_type]);
            flv->new_extradata_size[stream_type] = 0;
        }
    }
    if (stream_type == FLV_STREAM_TYPE_AUDIO &&
                    (sample_rate != flv->last_sample_rate ||
                     channels    != flv->last_channels)) {
        flv->last_sample_rate = sample_rate;
        flv->last_channels    = channels;
        ff_add_param_change(pkt, channels, 0, sample_rate, 0, 0);
    }
    //标记上Keyframe
    if (    stream_type == FLV_STREAM_TYPE_AUDIO ||
            ((flags & FLV_VIDEO_FRAMETYPE_MASK) == FLV_FRAME_KEY) ||
            stream_type == FLV_STREAM_TYPE_DATA)
        pkt->flags |= AV_PKT_FLAG_KEY;

leave:
    avio_skip(s->pb, 4);
    return ret;
}
```

flv_read_packet()的代码比较长，但是逻辑比较简单。它的主要功能就是根据FLV文件格式的规范，逐层解析Tag以及TagData，获取Tag以及TagData中的信息。比较关键的地方已经写上了注释，不再详细叙述。

parse_packet()

parse_packet()给需要AVCodecParser的媒体流提供解析AVPacket的功能。它的代码如下所示：

[cpp]

```
/**
 * Parse a packet, add all split parts to parse_queue.
 *
 * @param pkt Packet to parse, NULL when flushing the parser at end of stream.
 */
static int parse_packet(AVFormatContext *s, AVPacket *pkt, int stream_index)
{
    AVPacket out_pkt = { 0 }, flush_pkt = { 0 };
    AVStream *st = s->streams[stream_index];
    uint8_t *data = pkt ? pkt->data : NULL;
    int size      = pkt ? pkt->size : 0;
    int ret = 0, got_output = 0;

    if (!pkt) {
        av_init_packet(&flush_pkt);
        pkt        = &flush_pkt;
        got_output = 1;
    } else if (!size && st->parser->flags & PARSER_FLAG_COMPLETE_FRAMES) {
        // preserve 0-size sync packets
        compute_pkt_fields(s, st, st->parser, pkt);
    }

    while (size > 0 || (pkt == &flush_pkt && got_output)) {
        int len;

        av_init_packet(&out_pkt);
        //解析
        len = av_parser_parse2(st->parser, st->codec,
                               &out_pkt.data, &out_pkt.size, data, size,
                               pkt->pts, pkt->dts, pkt->pos);

        pkt->pts = pkt->dts = AV_NOPTS_VALUE;
        pkt->pos = -1;
        /* increment read pointer */
        data += len;
        size -= len;

        got_output = !!out_pkt.size;
        //继续
        if (!out_pkt.size)
            continue;

        if (pkt->side_data) {
            out_pkt.side_data       = pkt->side_data;
            out_pkt.side_data_elems = pkt->side_data_elems;
            pkt->side_data          = NULL;
            pkt->side_data_elems    = 0;
        }

        /* set the duration */
        out_pkt.duration = 0;
        if (st->codec->codec_type == AVMEDIA_TYPE_AUDIO) {
            if (st->codec->sample_rate > 0) {
                out_pkt.duration =
                    av_rescale_q_rnd(st->parser->duration,
                                     (AVRational) { 1, st->codec->sample_rate },
                                     st->time_base,
                                     AV_ROUND_DOWN);
            }
        }
        //设置属性值
        out_pkt.stream_index = st->index;
        out_pkt.pts          = st->parser->pts;
        out_pkt.dts          = st->parser->dts;
        out_pkt.pos          = st->parser->pos;

        if (st->need_parsing == AVSTREAM_PARSE_FULL_RAW)
            out_pkt.pos = st->parser->frame_offset;

        if (st->parser->key_frame == 1 ||
            (st->parser->key_frame == -1 &&
             st->parser->pict_type == AV_PICTURE_TYPE_I))
            out_pkt.flags |= AV_PKT_FLAG_KEY;

        if (st->parser->key_frame == -1 && st->parser->pict_type ==AV_PICTURE_TYPE_NONE && (pkt->flags&AV_PKT_FLAG_KEY))
            out_pkt.flags |= AV_PKT_FLAG_KEY;

        compute_pkt_fields(s, st, st->parser, &out_pkt);

        if (out_pkt.data == pkt->data && out_pkt.size == pkt->size) {
            out_pkt.buf = pkt->buf;
            pkt->buf    = NULL;
#if FF_API_DESTRUCT_PACKET
FF_DISABLE_DEPRECATION_WARNINGS
            out_pkt.destruct = pkt->destruct;
            pkt->destruct = NULL;
FF_ENABLE_DEPRECATION_WARNINGS
#endif
        }
        if ((ret = av_dup_packet(&out_pkt)) < 0)
            goto fail;

        if (!add_to_pktbuf(&s->parse_queue, &out_pkt, &s->parse_queue_end)) {
            av_free_packet(&out_pkt);
            ret = AVERROR(ENOMEM);
            goto fail;
        }
    }

    /* end of the stream => close and free the parser */
    if (pkt == &flush_pkt) {
        av_parser_close(st->parser);
        st->parser = NULL;
    }

fail:
    av_free_packet(pkt);
    return ret;
}
```

从代码中可以看出，最终调用了相应AVCodecParser的av_parser_parse2()函数，解析出来AVPacket。此后根据解析的信息还进行了一系列的赋值工作，不再详细叙述。

文章标签： ffmpeg 源代码 av_read_frame

个人分类： FFMPEG

所属专栏： 开源多媒体项目源代码分析 FFmpeg