**音视频编解码基础知识**

RGB、YUV像素数据处理

YUV 420 大小： w\*h\*3/2 Y：w\*h U：w\*h/4 V：w\*h/4

YUV 444 大小： w\*h\*3

YUV420P 大小：

1. 分离YUV420数据

for(int i=0;i<num;i++){

fread(pic,1,w\*h\*3/2,fp);

//Y

fwrite(pic,1,w\*h,fp1);

//U

fwrite(pic+w\*h,1,w\*h/4,fp2);

//V

fwrite(pic+w\*h\*5/4,1,w\*h/4,fp3);

}

1. 将YUV420P像素数据去掉颜色（变成灰度图）

如果想把 YUV 格式像素数据变成灰度图像，只需要将U、V分量设置成 128 即可。这是因为U、V是图像中的经过偏置处理的色度分量。色度分量在偏置处理前的取值范围是-128至127，这时候的无色对应的是“0”值。经过偏置后色度分量取值变成了0至255，因而此时的无色对应的就是128了。

1. 将YUV420P像素数据的亮度减半

Y 值变为之前的一半

1. 将YUV420P像素数据的周围加上边框

将边框范围内的数据，亮度设置为 255。

1. 计算两个YUV420P像素数据的PSNR

PSNR是最基本的视频质量评价方法。

mse\_sum+=pow((pic1[j]-pic2[j]),2);

mse = mse\_sum /(w\*h);

psnr = 10\*log10(255.0\*255.0/mse);

对于 8bit 量化的像素数据来说，PSNR 的计算公式：

https://img-blog.csdn.net/20160117233527240

上述公式中mse的计算公式如下所示。

https://img-blog.csdn.net/20160117233543104

其中M，N分别为图像的宽高，xij和yij分别为两张图像的每一个像素值。PSNR通常用于质量评价，就是计算受损图像与原始图像之间的差别，以此来评价受损图像的质量。

PSNR取值通常情况下都在**20-50**的范围内，取值越高，代表两张图像越接近，反映出受损图像质量越好。

1. 分离RGB24 的R、G、B等元素

for(int j=0;j<w\*h\*3;j=j+3){

//R

fwrite(pic+j,1,1,fp1);

//G

fwrite(pic+j+1,1,1,fp2);

//B

fwrite(pic+j+2,1,1,fp3);

}

RGB24格式的每个像素的三个分量是连续存储的。一帧宽高分别为w、h的RGB24图像一共占用w\*h\*3 Byte的存储空间。RGB24格式规定首先存储第一个像素的R、G、B，然后存储第二个像素的R、G、B…以此类推。类似于YUV420P的存储方式称为Planar方式，而类似于RGB24的存储方式称为Packed方式。

1. 将RGB24格式像素数据封装为BMP图像

1)将RGB数据前面加上文件头。

2)将RGB数据中每个像素的“B”和“R”的位置互换。

BMP文件是由BITMAPFILEHEADER、BITMAPINFOHEADER、RGB像素数据共3个部分构成，它的结构如下图所示。



BMP采用的是小端（Little Endian）存储方式。这种存储方式中“RGB24”格式的像素的分量存储的先后顺序为B、G、R。由于RGB24格式存储的顺序是R、G、B，所以需要将“R”和“B”顺序作一个调换再进行存储。

BMP文件头

typedef struct tagBITMAPFILEHEADER

{

unsigned short int bfType; //位图文件的类型，必须为BM

unsigned long bfSize; //文件大小，以字节为单位

unsigned short int bfReserverd1; //位图文件保留字，必须为0

unsigned short int bfReserverd2; //位图文件保留字，必须为0

unsigned long bfbfOffBits; //位图文件头到数据的偏移量，以字节为单位

}BITMAPFILEHEADER;

typedef struct tagBITMAPINFOHEADER

{

long biSize; //该结构大小，字节为单位

long biWidth; //图形宽度以象素为单位

long biHeight; //图形高度以象素为单位

short int biPlanes; //目标设备的级别，必须为1

short int biBitcount; //颜色深度，每个象素所需要的位数

short int biCompression; //位图的压缩类型

long biSizeImage; //位图的大小，以字节为单位

long biXPelsPermeter; //位图水平分辨率，每米像素数

long biYPelsPermeter; //位图垂直分辨率，每米像素数

long biClrUsed; //位图实际使用的颜色表中的颜色数

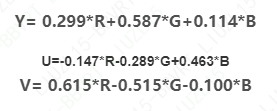
long biClrImportant; //位图显示过程中重要的颜色数

}BITMAPINFOHEADER;

BMP采用的是小端（Little Endian）存储方式。这种存储方式中“RGB24”格式的像素的分量存储的先后顺序为B、G、R。由于RGB24格式存储的顺序是R、G、B，所以需要将“R”和“B”顺序作一个调换再进行存储。

1. 将RGB24格式像素数据转换为YUV420P格式像素数据

RGB到YUV的转换公式：



1、RGB24存储方式是Packed，YUV420P存储方式是Packed。

2、U，V在水平和垂直方向的取样数是Y的一半

for (int j = 0; j<h;j++){

ptrRGB = RgbBuf + w\*j\*3 ;

for (int i = 0;i<w;i++){

r = \*(ptrRGB++);

g = \*(ptrRGB++);

b = \*(ptrRGB++);

y = (unsigned char)( ( 66 \* r + 129 \* g + 25 \* b + 128) >> 8) + 16 ;

u = (unsigned char)( ( -38 \* r - 74 \* g + 112 \* b + 128) >> 8) + 128 ;

v = (unsigned char)( ( 112 \* r - 94 \* g - 18 \* b + 128) >> 8) + 128 ;

\*(ptrY++) = clip\_value(y,0,255);

if (j%2==0&&i%2 ==0){

\*(ptrU++) =clip\_value(u,0,255);

}

else{

if (i%2==0){

\*(ptrV++) =clip\_value(v,0,255);

}

}

}

}

PCM音频数据处理

16LE。“16”代表采样位数是16bit。由于1Byte=8bit，所以一个声道的一个采样值占用2Byte。“LE”代表Little Endian，代表2 Byte采样值的存储方式为高位存在高地址中。

1. 分离PCM16LE双声道音频采样数据的左声道和右声道

while(!feof(fp)){

fread(sample,1,4,fp);

//L

fwrite(sample,1,2,fp1);

//R

fwrite(sample+2,1,2,fp2);

}

CM16LE双声道数据中左声道和右声道的采样值是间隔存储的。每个采样值占用2Byte空间。

1. 将PCM16LE双声道音频采样数据中左声道的音量降一半

unsigned char \*sample=(unsigned char \*)malloc(4);

while(!feof(fp)){

short \*samplenum=NULL;

fread(sample,1,4,fp);

samplenum=(short \*)sample; //读取了4个字节，但是 short 是2个字节，强制类型转换了

\*samplenum=\*samplenum/2;

//L

fwrite(sample,1,2,fp1);

//R

fwrite(sample+2,1,2,fp1);

cnt++;

}

读出左声道的2 Byte的取样值之后，将其当成了C语言中的一个short类型的变量。将该数值除以2之后写回到了PCM文件中。

1. 将PCM16LE双声道音频采样数据的声音速度提高一倍

unsigned char \*sample=(unsigned char \*)malloc(4);

while(!feof(fp)){

fread(sample,1,4,fp);

if(cnt%2!=0){

//L

fwrite(sample,1,2,fp1);

//R

fwrite(sample+2,1,2,fp1);

}

cnt++;

}

只采样了每个声道奇数点的样值。处理完成后，原本22秒左右的音频变成了11秒左右。音频的播放速度提高了2倍，音频的音调也变高了很多。

1. 将PCM16LE双声道音频采样数据转换为PCM8音频采样数据

unsigned char \*sample=(unsigned char \*)malloc(4);

while(!feof(fp)){

short \*samplenum16=NULL;

char samplenum8=0;

unsigned char samplenum8\_u=0;

fread(sample,1,4,fp);

//(-32768-32767)

samplenum16=(short \*)sample;

samplenum8=(\*samplenum16)>>8;

//(0-255)

samplenum8\_u=samplenum8+128;

//L

fwrite(&samplenum8\_u,1,1,fp1);

samplenum16=(short \*)(sample+2);

samplenum8=(\*samplenum16)>>8;

samplenum8\_u=samplenum8+128;

//R

fwrite(&samplenum8\_u,1,1,fp1);

cnt++;

}

PCM16LE格式的采样数据的取值范围是-32768到32767，而PCM8格式的采样数据的取值范围是0到255。所以PCM16LE转换到PCM8需要经过两个步骤：第一步是将-32768到32767的16bit有符号数值转换为-128到127的8bit有符号数值，第二步是将-128到127的8bit有符号数值转换为0到255的8bit无符号数值。

1. 将从PCM16LE单声道音频采样数据中截取一部分数据

unsigned char \*sample=(unsigned char \*)malloc(2);

int cnt=0;

while(!feof(fp)){

fread(sample,1,2,fp);

if(cnt>start\_num&&cnt<=(start\_num+dur\_num)){

fwrite(sample,1,2,fp1);

}

cnt++;

}

1. 将PCM16LE双声道音频采样数据转换为WAVE格式音频数据

WAVE格式音频（扩展名为“.wav”）是Windows系统中最常见的一种音频。该格式的实质就是在PCM文件的前面加了一个文件头。

typedef struct WAVE\_HEADER{

char fccID[4];

unsigned long dwSize;

char fccType[4];

}WAVE\_HEADER;

typedef struct WAVE\_FMT{

char fccID[4];

unsigned long dwSize;

unsigned short wFormatTag;

unsigned short wChannels;

unsigned long dwSamplesPerSec;

unsigned long dwAvgBytesPerSec;

unsigned short wBlockAlign;

unsigned short uiBitsPerSample;

}WAVE\_FMT;

typedef struct WAVE\_DATA{

char fccID[4];

unsigned long dwSize;

}WAVE\_DATA;

通过在PCM文件前面加一个WAVE文件头从而封装为WAVE格式音频。

//WAVE\_HEADER

memcpy(pcmHEADER.fccID,"RIFF",strlen("RIFF"));

memcpy(pcmHEADER.fccType,"WAVE",strlen("WAVE"));

fseek(fpout,sizeof(WAVE\_HEADER),1);

//WAVE\_FMT

pcmFMT.dwSamplesPerSec=sample\_rate;

pcmFMT.dwAvgBytesPerSec=pcmFMT.dwSamplesPerSec\*sizeof(m\_pcmData);

pcmFMT.uiBitsPerSample=bits;

memcpy(pcmFMT.fccID,"fmt ",strlen("fmt "));

pcmFMT.dwSize=16;

pcmFMT.wBlockAlign=2;

pcmFMT.wChannels=channels;

pcmFMT.wFormatTag=1;

fwrite(&pcmFMT,sizeof(WAVE\_FMT),1,fpout);

//WAVE\_DATA;

memcpy(pcmDATA.fccID,"data",strlen("data"));

pcmDATA.dwSize=0;

fseek(fpout,sizeof(WAVE\_DATA),SEEK\_CUR);

fread(&m\_pcmData,sizeof(unsigned short),1,fp);

while(!feof(fp)){

pcmDATA.dwSize+=2;

fwrite(&m\_pcmData,sizeof(unsigned short),1,fpout);

fread(&m\_pcmData,sizeof(unsigned short),1,fp);

}

pcmHEADER.dwSize=44+pcmDATA.dwSize;

rewind(fpout);

fwrite(&pcmHEADER,sizeof(WAVE\_HEADER),1,fpout);

fseek(fpout,sizeof(WAVE\_FMT),SEEK\_CUR);

fwrite(&pcmDATA,sizeof(WAVE\_DATA),1,fpout);

WAVE文件是一种RIFF格式的文件。其基本块名称是“WAVE”，其中包含了两个子块“fmt”和“data”。从编程的角度简单说来就是由WAVE\_HEADER、WAVE\_FMT、WAVE\_DATA、采样数据共4个部分组成。它的结构如下所示：



有一点需要注意：WAVE\_HEADER和WAVE\_DATA中包含了一个文件长度信息的dwSize字段，该字段的值必须在写入完音频采样数据之后才能获得。因此这两个结构体最后才写入WAVE文件中。

H.264视频码流解析

H.264原始码流（又称为“裸流”）是由一个一个的NALU组成的。他们的结构如下图所示：

https://img-blog.csdn.net/20160118001549018

其中每个NALU之间通过startcode（起始码）进行分隔，起始码分成两种：0x000001（3Byte）或者0x00000001（4Byte）。如果NALU对应的Slice为一帧的开始就用0x00000001，否则就用0x000001。

H.264码流解析的步骤就是**首先从码流中搜索0x000001和0x00000001，分离出NALU**；然后再分析NALU的各个字段。

typedef enum {

NALU\_TYPE\_SLICE = 1,

NALU\_TYPE\_DPA = 2,

NALU\_TYPE\_DPB = 3,

NALU\_TYPE\_DPC = 4,

NALU\_TYPE\_IDR = 5,

NALU\_TYPE\_SEI = 6,

NALU\_TYPE\_SPS = 7,

NALU\_TYPE\_PPS = 8,

NALU\_TYPE\_AUD = 9,

NALU\_TYPE\_EOSEQ = 10,

NALU\_TYPE\_EOSTREAM = 11,

NALU\_TYPE\_FILL = 12,

} NaluType;

typedef enum {

NALU\_PRIORITY\_DISPOSABLE = 0,

NALU\_PRIRITY\_LOW = 1,

NALU\_PRIORITY\_HIGH = 2,

NALU\_PRIORITY\_HIGHEST = 3

} NaluPriority;

typedef struct

{

int startcodeprefix\_len;

//! 4 for parameter sets and first slice in picture, 3 for everything else (suggested)

unsigned len;

//! Length of the NAL unit (Excluding the start code, which does not belong to the NALU)

unsigned max\_size; //! Nal Unit Buffer size

int forbidden\_bit; //! should be always FALSE

int nal\_reference\_idc; //! NALU\_PRIORITY\_xxxx

int nal\_unit\_type; //! NALU\_TYPE\_xxxx

char \*buf; //! contains the first byte followed by the EBSP

} NALU\_t;

//寻找 NALU 的起始码

static int FindStartCode2 (unsigned char \*Buf){

if(Buf[0]!=0 || Buf[1]!=0 || Buf[2] !=1) return 0; //0x000001?

else return 1;

}

static int FindStartCode3 (unsigned char \*Buf){

if(Buf[0]!=0 || Buf[1]!=0 || Buf[2] !=0 || Buf[3] !=1) return 0;//0x00000001?

else return 1;

}

int GetAnnexbNALU (NALU\_t \*nalu){

int pos = 0;

int StartCodeFound, rewind;

unsigned char \*Buf;

if ((Buf = (unsigned char\*)calloc (nalu->max\_size , sizeof(char))) == NULL)

printf ("GetAnnexbNALU: Could not allocate Buf memory\n");

nalu->startcodeprefix\_len=3;

if (3 != fread (Buf, 1, 3, h264bitstream)){

free(Buf);

return 0;

}

info2 = FindStartCode2 (Buf);

if(info2 != 1) {

if(1 != fread(Buf+3, 1, 1, h264bitstream)){

free(Buf);

return 0;

}

info3 = FindStartCode3 (Buf);

if (info3 != 1){

free(Buf);

return -1;

}

else {

pos = 4;

nalu->startcodeprefix\_len = 4;

}

}

else{

nalu->startcodeprefix\_len = 3;

pos = 3;

}

StartCodeFound = 0;

info2 = 0;

info3 = 0;

while (!StartCodeFound){

if (feof (h264bitstream)){

nalu->len = (pos-1)-nalu->startcodeprefix\_len;

memcpy (nalu->buf, &Buf[nalu->startcodeprefix\_len], nalu->len);

nalu->forbidden\_bit = nalu->buf[0] & 0x80; //1 bit

nalu->nal\_reference\_idc = nalu->buf[0] & 0x60; // 2 bit

nalu->nal\_unit\_type = (nalu->buf[0]) & 0x1f;// 5 bit

free(Buf);

return pos-1;

}

Buf[pos++] = fgetc (h264bitstream);

info3 = FindStartCode3(&Buf[pos-4]);

if(info3 != 1)

info2 = FindStartCode2(&Buf[pos-3]);

StartCodeFound = (info2 == 1 || info3 == 1);

}

// Here, we have found another start code (and read length of startcode bytes more than we should

// have. Hence, go back in the file

rewind = (info3 == 1)? -4 : -3;

if (0 != fseek (h264bitstream, rewind, SEEK\_CUR)){

free(Buf);

printf("GetAnnexbNALU: Cannot fseek in the bit stream file");

}

// Here the Start code, the complete NALU, and the next start code is in the Buf.

// The size of Buf is pos, pos+rewind are the number of bytes excluding the next

// start code, and (pos+rewind)-startcodeprefix\_len is the size of the NALU excluding the start code

nalu->len = (pos+rewind)-nalu->startcodeprefix\_len;

memcpy (nalu->buf, &Buf[nalu->startcodeprefix\_len], nalu->len);//

nalu->forbidden\_bit = nalu->buf[0] & 0x80; //1 bit

nalu->nal\_reference\_idc = nalu->buf[0] & 0x60; // 2 bit

nalu->nal\_unit\_type = (nalu->buf[0]) & 0x1f;// 5 bit

free(Buf);

return (pos+rewind);

}

int simplest\_h264\_parser(char \*url){

NALU\_t \*n;

int buffersize=100000;

//FILE \*myout=fopen("output\_log.txt","wb+");

FILE \*myout=stdout;

h264bitstream=fopen(url, "rb+");

if (h264bitstream==NULL){

printf("Open file error\n");

return 0;

}

n = (NALU\_t\*)calloc (1, sizeof (NALU\_t));

if (n == NULL){

printf("Alloc NALU Error\n");

return 0;

}

n->max\_size=buffersize;

n->buf = (char\*)calloc (buffersize, sizeof (char));

if (n->buf == NULL){

free (n);

printf ("AllocNALU: n->buf");

return 0;

}

int data\_offset=0;

int nal\_num=0;

printf("-----+-------- NALU Table ------+---------+\n");

printf(" NUM | POS | IDC | TYPE | LEN |\n");

printf("-----+---------+--------+-------+---------+\n");

while(!feof(h264bitstream))

{

int data\_lenth;

data\_lenth=GetAnnexbNALU(n);

char type\_str[20]={0};

switch(n->nal\_unit\_type){

case NALU\_TYPE\_SLICE:sprintf(type\_str,"SLICE");break;

case NALU\_TYPE\_DPA:sprintf(type\_str,"DPA");break;

case NALU\_TYPE\_DPB:sprintf(type\_str,"DPB");break;

case NALU\_TYPE\_DPC:sprintf(type\_str,"DPC");break;

case NALU\_TYPE\_IDR:sprintf(type\_str,"IDR");break;

case NALU\_TYPE\_SEI:sprintf(type\_str,"SEI");break;

case NALU\_TYPE\_SPS:sprintf(type\_str,"SPS");break;

case NALU\_TYPE\_PPS:sprintf(type\_str,"PPS");break;

case NALU\_TYPE\_AUD:sprintf(type\_str,"AUD");break;

case NALU\_TYPE\_EOSEQ:sprintf(type\_str,"EOSEQ");break;

case NALU\_TYPE\_EOSTREAM:sprintf(type\_str,"EOSTREAM");break;

case NALU\_TYPE\_FILL:sprintf(type\_str,"FILL");break;

}

char idc\_str[20]={0};

switch(n->nal\_reference\_idc>>5){

case NALU\_PRIORITY\_DISPOSABLE:sprintf(idc\_str,"DISPOS");break;

case NALU\_PRIRITY\_LOW:sprintf(idc\_str,"LOW");break;

case NALU\_PRIORITY\_HIGH:sprintf(idc\_str,"HIGH");break;

case NALU\_PRIORITY\_HIGHEST:sprintf(idc\_str,"HIGHEST");break;

}

fprintf(myout,"%5d| %8d| %7s| %6s| %8d|\n",nal\_num,data\_offset,idc\_str,type\_str,n->len);

data\_offset=data\_offset+data\_lenth;

nal\_num++;

}

//Free

if (n){

if (n->buf){

free(n->buf);

n->buf=NULL;

}

free (n);

}

return 0;

}

AAC音频码流解析

AAC原始码流（又称为“裸流”）是由一个一个的ADTS frame组成的。他们的结构如下图所示。

https://img-blog.csdn.net/20160118101611729

其中每个ADTS frame之间通过syncword（同步字）进行分隔。同步字为0xFFF（二进制“111111111111”）。AAC码流解析的步骤就是首先从码流中搜索0x0FFF，分离出ADTS frame；然后再分析ADTS frame的首部各个字段。

首部各个字段：（0~6 共7字节）

* 1. : AAC码流标识

2：

最高两位表示协议，有 Main、LC、SSR等

后6位的4倍表示采样率，如：44100HZ

3-5：size

6：暂不知

#include <stdio.h>

#include <stdlib.h>

#include <string.h>

int getADTSframe(unsigned char\* buffer, int buf\_size, unsigned char\* data ,int\* data\_size){

int size = 0;

if(!buffer || !data || !data\_size ){

return -1;

}

while(1){

if(buf\_size < 7 ){

return -1;

}

//Sync words

if((buffer[0] == 0xff) && ((buffer[1] & 0xf0) == 0xf0) ){

size |= ((buffer[3] & 0x03) <<11); //high 2 bit

size |= buffer[4]<<3; //middle 8 bit

size |= ((buffer[5] & 0xe0)>>5); //low 3bit

break;

}

--buf\_size;

++buffer;

}

if(buf\_size < size){

return 1;

}

memcpy(data, buffer, size);

\*data\_size = size;

return 0;

}

int simplest\_aac\_parser(char \*url)

{

int data\_size = 0;

int size = 0;

int cnt=0;

int offset=0;

//FILE \*myout=fopen("output\_log.txt","wb+");

FILE \*myout=stdout;

unsigned char \*aacframe=(unsigned char \*)malloc(1024\*5);

unsigned char \*aacbuffer=(unsigned char \*)malloc(1024\*1024);

FILE \*ifile = fopen(url, "rb");

if(!ifile){

printf("Open file error");

return -1;

}

printf("-----+- ADTS Frame Table -+------+\n");

printf(" NUM | Profile | Frequency| Size |\n");

printf("-----+---------+----------+------+\n");

while(!feof(ifile)){

data\_size = fread(aacbuffer+offset, 1, 1024\*1024-offset, ifile);

unsigned char\* input\_data = aacbuffer;

while(1)

{

int ret=getADTSframe(input\_data, data\_size, aacframe, &size);

if(ret==-1){

break;

}else if(ret==1){

memcpy(aacbuffer,input\_data,data\_size);

offset=data\_size;

break;

}

char profile\_str[10]={0};

char frequence\_str[10]={0};

unsigned char profile=aacframe[2]&0xC0;

profile=profile>>6;

switch(profile){

case 0: sprintf(profile\_str,"Main");break;

case 1: sprintf(profile\_str,"LC");break;

case 2: sprintf(profile\_str,"SSR");break;

default:sprintf(profile\_str,"unknown");break;

}

unsigned char sampling\_frequency\_index=aacframe[2]&0x3C;

sampling\_frequency\_index=sampling\_frequency\_index>>2;

switch(sampling\_frequency\_index){

case 0: sprintf(frequence\_str,"96000Hz");break;

case 1: sprintf(frequence\_str,"88200Hz");break;

case 2: sprintf(frequence\_str,"64000Hz");break;

case 3: sprintf(frequence\_str,"48000Hz");break;

case 4: sprintf(frequence\_str,"44100Hz");break;

case 5: sprintf(frequence\_str,"32000Hz");break;

case 6: sprintf(frequence\_str,"24000Hz");break;

case 7: sprintf(frequence\_str,"22050Hz");break;

case 8: sprintf(frequence\_str,"16000Hz");break;

case 9: sprintf(frequence\_str,"12000Hz");break;

case 10: sprintf(frequence\_str,"11025Hz");break;

case 11: sprintf(frequence\_str,"8000Hz");break;

default:sprintf(frequence\_str,"unknown");break;

}

fprintf(myout,"%5d| %8s| %8s| %5d|\n",cnt,profile\_str ,frequence\_str,size);

data\_size -= size;

input\_data += size;

cnt++;

}

}

fclose(ifile);

free(aacbuffer);

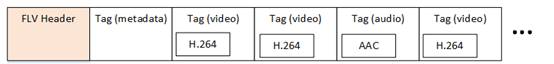
free(aacframe);

return 0;

}

FLV封装格式解析

FLV封装格式是由一个FLV Header文件头和一个一个的Tag组成的。Tag中包含了音频数据以及视频数据。FLV的结构如下图所示。



FlvHeader:

struct{

byte Signature[3];

byte Version;

byte Flags;

uint DataOffsets;

}FLV\_HEADER;

TagHeader:

typedef struct{

byte TagType;

byte DataSize[3];

byte Timestamp[3];

uint Reserved;

}

TagType:

TAG\_TYPE\_AUDIO

TAG\_TYPE\_VIDEO

TAG\_TYPE\_SCRIPT

DataSize： DataSize[0]\*65536+ DataSize[1]\*256+ DataSize[2]

Timestamp[3]：Timestamp[0]\*65536+ Timestamp[1]\*256+ Timestamp[2]

数据首字节：

TagType：

TAG\_TYPE\_ADUIO:

0：声道

1：位数

2-3：采样率

4-7：协议格式

TAG\_TYPE\_VIDEO:

0-3位：

0： UNKNOWN

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

4-7位：

1：key frame

2：inter frame

3：disposable inter frame

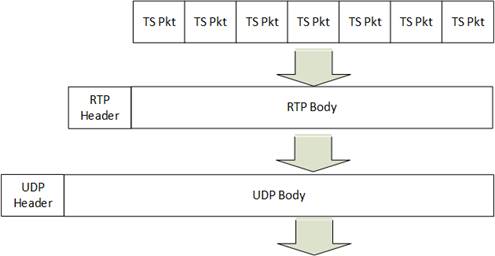
4：generated keyframe

5：video info/command frame

other：UNKNOWN

UDP-RTP协议解析：

首先每7个MPEG-TS Packet打包为一个RTP，然后每个RTP再打包为一个UDP。其中打包RTP的方法就是在MPEG-TS数据前面加上RTP Header，而打包RTP的方法就是在RTP数据前面加上UDP Header。



typedef struct RTP\_FIXED\_HEADER{

/\* byte 0 \*/

unsigned char csrc\_len:4; /\* expect 0 \*/

unsigned char extension:1; /\* expect 1 \*/

unsigned char padding:1; /\* expect 0 \*/

unsigned char version:2; /\* expect 2 \*/

/\* byte 1 \*/

unsigned char payload:7;

unsigned char marker:1; /\* expect 1 \*/

/\* bytes 2, 3 \*/

unsigned short seq\_no;

/\* bytes 4-7 \*/

unsigned long timestamp;

/\* bytes 8-11 \*/

unsigned long ssrc; /\* stream number is used here. \*/

} RTP\_FIXED\_HEADER;

typedef struct MPEGTS\_FIXED\_HEADER {

unsigned sync\_byte: 8;

unsigned transport\_error\_indicator: 1;

unsigned payload\_unit\_start\_indicator: 1;

unsigned transport\_priority: 1;

unsigned PID: 13;

unsigned scrambling\_control: 2;

unsigned adaptation\_field\_exist: 2;

unsigned continuity\_counter: 4;

} MPEGTS\_FIXED\_HEADER;

RTP:

payload:

0-18: "Audio"

31: "H.261"

32: "MPV"

33: "MP2T"

34: "H.263"

96: "H.264"

default:"other"

MPEGTS:

起始字节为 0x47

长度为 188