* **webrtc/common\_types.h**
* **webrtc/modules/interface/module\_common\_types.h**
* **webrtc/modules/rtp\_rtcp/source/rtp\_utility.h**
* **webrtc/modules/rtp\_rtcp/source/rtp\_utility.cc**
* **webrtc/modules/rtp\_rtcp/source/rtp\_sender\_video.h**
* **webrtc/modules/rtp\_rtcp/source/rtp\_sender\_video.cc**
* **webrtc/modules/rtp\_rtcp/source/rtp\_receiver\_video.h**
* **webrtc/modules/rtp\_rtcp/source/rtp\_receiver\_video.cc**
* **webrtc/modules/rtp\_rtcp/source/rtp\_payload\_registry.cc**
* **webrtc/modules/video\_coding/codecs/interface/video\_codec\_interface.h**
* **webrtc/modules/video\_coding/main/interface/video\_coding\_defines.h**
* **webrtc/modules/video\_coding/main/source/internal\_defines.h**
* **webrtc/modules/video\_coding/main/source/codec\_database.cc**
* **webrtc/modules/video\_coding/main/source/decoding\_state.cc (\* v2)**
* **webrtc/modules/video\_coding/main/source/packet.cc**
* **webrtc/modules/video\_coding/main/source/encoded\_frame.cc**
* **webrtc/modules/video\_coding/main/source/frame\_buffer.cc**
* **webrtc/modules/video\_coding/main/source/session\_info.h**
* **webrtc/modules/video\_coding/main/source/session\_info.cc**
* **webrtc/modules/video\_coding/main/source/generic\_encoder.cc**
* **webrtc/video\_engine/vie\_channel.cc**
* **webrtc/video\_engine/vie\_codec\_impl.cc**
* **webrtc/video\_engine/vie\_encoder.cc**
* **talk/media/webrtc/webrtcvideoengine.cc (\* v2)**
* **talk/app/webrtc/jsepsessiondescription.cc 设置sdp默认值,未启用**

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**webrtc/common defines**

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* **webrtc/common\_types.h**

**+48~73**

添加定义：

VMXMAKEFOURCC

VMXFOURCC

fourccH264

fourccVP8

struct VideoCodecVMX

**+633**

enum VideoCodecType 添加kVideoCodecVMX

**+640**

union VideoCodecUnion添加VideoCodecVMX vmx

* **webrtc/modules/interface/module\_common\_types.h**

**+72~76**

定义enum RtpPayloadType

**+78~86**

定义struct RTPVideoHeaderH264

**+116**

union RTPVideoTypeHeader添加RTPVideoHeaderH264 H264

**+123**

enum RtpVideoCodecTypes添加kRtpVideoH264

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**webrtc/modules/rtp\_rtcp**

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* **webrtc/modules/rtp\_rtcp/source/rtp\_utility.h**

**+149~155**  
定义struct RTPPayloadH264  
**+160**  
union RTPPayloadUnion添加RTPPayloadH264 H264  
**+190**  
添加bool ParseH264 (RTPPayload & parsedPacket) const;

* **webrtc/modules/rtp\_rtcp/source/rtp\_utility.cc**

**+213~215**  
RTPPayload::SetType 添加264的判断  
**+581~582**  
RTPPayloadParser::Parse 添加对于264的判断  
**+592~616**  
添加bool RTPPayloadParser::ParseH264( RTPPayload & parsedPacket)

* **webrtc/modules/rtp\_rtcp/source/rtp\_sender\_video.h**

**+114~121**

添加SendH264()

* **webrtc/modules/rtp\_rtcp/source/rtp\_sender\_video.cc**

**+97~99**  
RTPSenderVideo::RegisterVideoPayload 添加264的注册  
**+321~330**  
RTPSenderVideo::SendVideo添加264的分支  
**+419~478**  
RTPSenderVideo::SendH264()添加264的发送处理

* **webrtc/modules/rtp\_rtcp/source/rtp\_receiver\_video.h**

**+68~71**

添加ReceiveH264Codec()

* **webrtc/modules/rtp\_rtcp/source/rtp\_receiver\_video.cc**

**+129~130**

RTPReceiverVideo::ParseVideoCodecSpecific添加264的判断分支

**+171~215**

添加RTPReceiverVideo::ReceiveH264Codec()264的接收处理

* **webrtc/modules/rtp\_rtcp/source/rtp\_payload\_registry.cc**

**+455~457**

RTPPayloadVideoStrategy::CreatePayloadType()添加264的判断

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**webrtc/modules/video\_coding**

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* **webrtc/modules/video\_coding/codecs/interface/video\_codec\_interface.h**

**+46~50**

定义struct CodecSpecificInfoH264

**+60**

union CodecSpecificInfoUnion添加CodecSpecificInfoH264 H264

* **webrtc/modules/video\_coding/main/interface/video\_coding\_defines.h**

**+43**  
定义VCM\_H264\_PAYLOAD\_TYPE  127

* **webrtc/modules/video\_coding/main/source/internal\_defines.h**

**+48**  
定义VCM\_H264\_IDX VCM\_I420\_IDX+1  
**-+49**  
定义VCM\_NUM\_VIDEO\_CODECS\_AVAILABLE VCM\_H264\_IDX + 1

* **webrtc/modules/video\_coding/main/source/codec\_database.cc**

**+579~582**

VCMCodecDataBase::SupportsRenderScheduling() 修改判断逻辑,添加NULL判断

**+126~141**

VCMCodecDataBase::Codec 添加H264默认编码参数

* **webrtc/modules/video\_coding/main/source/decoding\_state.cc**

VCMDecodingState::ContinuousFrame 连续frame判断条件有待商榷

* **webrtc/modules/video\_coding/main/source/packet.cc**

**+114~127**

VCMPacket::CopyCodecSpecifics 从rtp头中提取264的信息到packet

* **webrtc/modules/video\_coding/main/source/encoded\_frame.cc**

**+139~148**

VCMEncodedFrame::CopyCodecSpecific 从rtp头中提取264的信息到encoderframe

* **webrtc/modules/video\_coding/main/source/frame\_buffer.cc**

**+160**

VCMFrameBuffer::InsertPacket 在一帧完成时是否设置帧类型,有待商榷,  
官方似乎已经在PrepareForDecode中修改了这个问题  
**-+274~303**

VCMFrameBuffer::PrepareForDecode()添加264分片信息的预处理

* **webrtc/modules/video\_coding/main/source/session\_info.h**

**+63~64**  
定义BuildH264FragmentationHeader()

* **webrtc/modules/video\_coding/main/source/session\_info.cc**

**+253~280**  
添加VCMSessionInfo::BuildH264FragmentationHeader()264分片预处理

* **webrtc/modules/video\_coding/main/source/generic\_encoder.cc**

**+264~269**

VCMEncodedFrameCallback::CopyCodecSpecific 将CodecSpecificInfo赋值到rtp头中

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**webrtc/video\_engine**

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* **webrtc/video\_engine/vie\_channel.cc**

**+221~237**

ViEChannel::Init 向VIEChannel添加264的编码参数

* **webrtc/video\_engine/vie\_codec\_impl.cc**

**+771~772**

ViECodecImpl::CodecValid 对H264编码类型名字的有效性判断

* **webrtc/video\_engine/vie\_encoder.cc**

待定

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**talk/**

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* **talk/media/webrtc/webrtcvideoengine.cc**

**-+99**

修改MTU默认值为1500

static const int kVideoMtu = 1500;

**+106**

添加H264编码名定义  
static const char kH264PayloadName[] = "H264";

**-+113**

修改外表payload基础值为127(120)  
static const int kExternalVideoPayloadTypeBase = 127;

**+761**

WebRtcVideoEngine::kVideoCodecPrefs添加预支持的编码类型

**+3290~3293**

WebRtcVideoMediaChannel::SetSendCodec 添加264的初始化