**Webrtc内部设计索引**

<http://www.webrtc.org/reference/webrtc-internals>

**(1)、网络传输模块：libjingle**

WebRTC重用了 libjingle的一些组件，主要是 network和 transport组件，关于 libjingle的文档资料可以查看 [**这里**](http://code.google.com/apis/talk/libjingle/developer_guide.html)。

**(2)、媒体处理的主要数据结构**

*注意：以下所有的方法、类、结构体、枚举常量等都在* *webrtc命名空间里*

|  |  |  |
| --- | --- | --- |
| **类、结构体、枚举常量** | **头文件** | **说明** |
| [**Structures**](http://www.webrtc.org/reference/webrtc-internals/structures) | common\_types.h | 列出VoiceEngine & VideoEngine的通用Struct |
| [**Enumerators**](http://www.webrtc.org/reference/webrtc-internals/enumerators) | common\_types.h | 列出VoiceEngine & VideoEngine的通用Enum |
| [**Classes**](http://www.webrtc.org/reference/webrtc-internals/classes) | common\_types.h | 列出VoiceEngine & VideoEngine的通用Class |
| class  **[VoiceEngine](http://www.webrtc.org/reference/webrtc-internals/voiceengine-1)** | voe\_base.h | 描述使用factory方法的VoiceEngine类如何分配和释放资源.也列出了一些APIs,调用它们可以将文件追踪当作回调消息.  It also lists the APIs which are required to enable file tracing and/or traces as callback messages |
| class  **[VideoEngine](http://www.webrtc.org/reference/webrtc-internals/videoengine)** | vie\_base.h | 描述使用factory方法的VideoEngine 类如何分配和释放资源.也列出了一些APIs,调用它们可以将文件追踪当作回调消息 |

**(3)、音频引擎（VoiceEngine** **）模块** **APIs**

*下表列的是目前在* *VoiceEngine中可用的sub APIs*

|  |  |  |
| --- | --- | --- |
| **sub-API** | **头文件** | **说明** |
| [**VoEAudioProcessing**](http://www.webrtc.org/reference/webrtc-internals/voeaudioprocessing) | voe\_audio\_processing.h | 添加支持噪音抑制(NS),自动扩音控制(AGC)和回音控制(EC).也包含了接收端的VAD(音频活跃检测) |
| [**VoEBase**](http://www.webrtc.org/reference/webrtc-internals/voebase) | voe\_base.h | 通过使用G711支持Voip的双向通信。  注：这个API必须始终被创建 |
| [**VoECallReport**](http://www.webrtc.org/reference/webrtc-internals/voecallreport) | voe\_call\_report.h | 添加支持了呼叫报告,其中包含了一些保活检测、  RTT测试和回应检测. |
| [**VoECodec**](http://www.webrtc.org/reference/webrtc-internals/voecodec) | voe\_codec.h | 添加了一些非默认编解码器(如iLBC, iSAC, G.722等)  和音频活跃度检测支持(VAD) |
| [**VoEDTMF**](http://www.webrtc.org/reference/webrtc-internals/voedtmf) | voe\_dtmf.h | Adds telephone event transmission, DTMF tone generation and telephone event detection. (Telephone events include DTMF.) |
| [**VoEEncryption**](http://www.webrtc.org/reference/webrtc-internals/voeencryption) | voe\_encryption.h | Adds external encryption/decryption support. |
| [**VoEErrors**](http://www.webrtc.org/reference/webrtc-internals/voeerrors) | voe\_errors.h | Error Codes for the VoiceEngine |
| [**VoEExternalMedia**](http://www.webrtc.org/reference/webrtc-internals/voeexternalmedia) | voe\_external\_media.h | Adds support for external media processing and enables utilization of an external audio resource. |
| [**VoEFile**](http://www.webrtc.org/reference/webrtc-internals/voefile) | voe\_file.h | Adds file playback, file recording and file conversion functions. |
| [**VoEHardware**](http://www.webrtc.org/reference/webrtc-internals/voehardware) | voe\_hardware.h | Adds sound device handling, CPU load monitoring and device information functions. |
| [**VoENetEqStats**](http://www.webrtc.org/reference/webrtc-internals/voeneteqstats) | voe\_neteq\_stats.h | Adds buffer statistics functions. |
| [**VoENetwork**](http://www.webrtc.org/reference/webrtc-internals/voenetwork) | voe\_network.h | Adds external transport, port and address filtering, Windows QoS support and packet timeout notifications. |
| [**VoERTP\_RTCP**](http://www.webrtc.org/reference/webrtc-internals/voertp_rtcp) | voe\_rtp\_rtcp.h | Adds support for RTCP sender reports, SSRC handling, RTP/RTCP statistics, Forward Error Correction (FEC), RTCP APP, RTP capturing and RTP keepalive. |
| [**VoEVideoSync**](http://www.webrtc.org/reference/webrtc-internals/voevideosync) | voe\_video\_sync.h | Adds RTP header modification support, playout-delay tuning and monitoring. |
| [**VoEVolumeControl**](http://www.webrtc.org/reference/webrtc-internals/voevolumecontrol) | voe\_volume\_control.h | Adds speaker volume controls, microphone volume controls, mute support, and additional stereo scaling methods. |

**(4)、视频引擎（VideoEngine）模块** **APIs**

*下表列的是目前在* *VideoEngine中可用的sub APIs*

|  |  |  |
| --- | --- | --- |
| **sub-API** | **头文件** | **说明** |
| [**ViEBase**](http://www.webrtc.org/reference/webrtc-internals/viebase) | vie\_base.h | Basic functionality for creating a VideoEngine instance, channels and VoiceEngine interaction.  **NOTE:** This API must always be created. |
| [**ViECapture**](http://www.webrtc.org/reference/webrtc-internals/viecapture) | vie\_capture.h | Adds support for capture device allocation as well as capture device capabilities. |
| [**ViECodec**](http://www.webrtc.org/reference/webrtc-internals/viecodec) | vie\_codec.h | Adds non-default codecs, codec settings and packet loss functionality. |
| [**ViEEncryption**](http://www.webrtc.org/reference/webrtc-internals/vieencryption) | vie\_encryption.h | Adds external encryption/decryption support. |
| [**ViEErrors**](http://www.webrtc.org/reference/webrtc-internals/vieexternalcodec) | vie\_errors.h | Error codes for the VideoEngine |
| [**ViEExternalCodec**](http://www.webrtc.org/reference/webrtc-internals/vieexternalcodec) | vie\_external\_codec.h | Adds support for using external codecs. |
| [**ViEFile**](http://www.webrtc.org/reference/webrtc-internals/viefile) | vie\_file.h | Adds support for file recording, file playout, background images and snapshot. |
| [**ViEImageProcess**](http://www.webrtc.org/reference/webrtc-internals/vieimageprocess) | vie\_image\_process.h | Adds effect filters, deflickering, denoising and color enhancement. |
| [**ViENetwork**](http://www.webrtc.org/reference/webrtc-internals/vienetwork) | vie\_network.h | Adds send and receive functionality, external transport, port and address filtering, Windows QoS support, packet timeout notification and changes to network settings. |
| [**ViERender**](http://www.webrtc.org/reference/webrtc-internals/vierender) | vie\_render.h | Adds rendering functionality. |
| [**ViERTP\_RTCP**](http://www.webrtc.org/reference/webrtc-internals/viertp_rtcp) | vie\_rtp\_rtcp.h | Adds support for RTCP reports, SSRS handling RTP/RTCP statistics, NACK/FEC, keep-alive functionality and key frame request methods. |