

原 RTMPdump 源代码分析 1：main()函数

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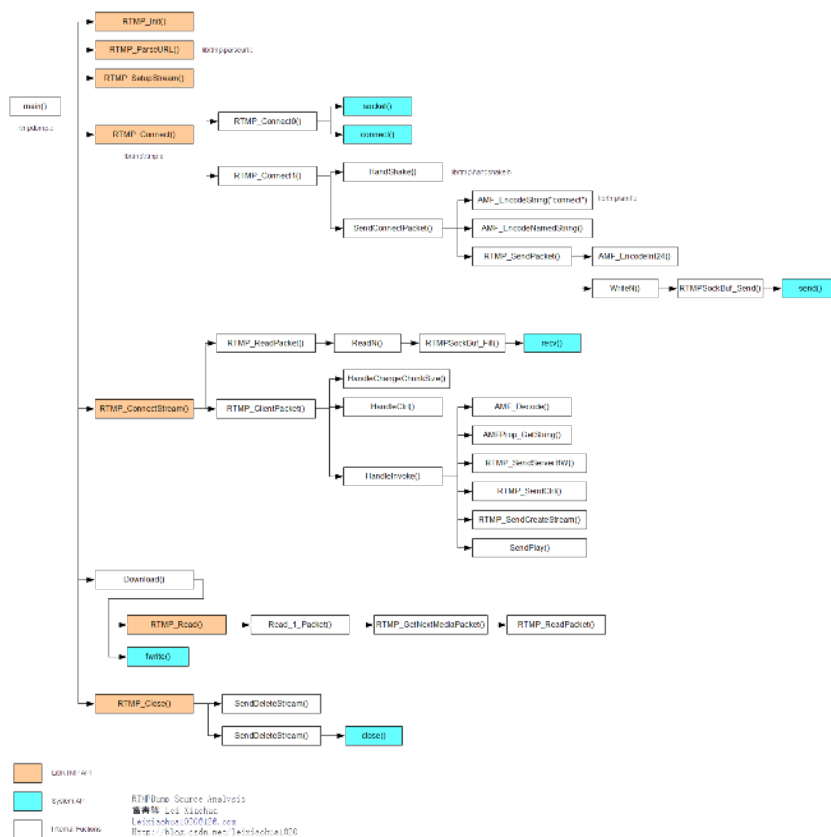
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rtmpdump 是一个用来处理 RTMP 流媒体的工具包,支持 rtmp://, rtmpt://, rtmpe://, rtmpte://, and rtmps:// 等。之前在学习RTMP协议的时候,发现没有讲它源代码的,只好自己分析,现在打算把自己学习的成果写出来,可能结果不一定都对,先暂且记录一下。

函数调用结构图

RTMPDump (libRTMP)的整体的函数调用结构图如下图所示。



[单击查看大图](#)

详细分析

使用RTMPdump下载一个流媒体的大致流程是这样的：

```
[cpp]
1.  RTMP_Init();//初始化结构体
2.  InitSockets();//初始化Socket
3.  RTMP_ParseURL();//解析输入URL
4.  RTMP_SetupStream();//一些设置
5.  fopen();//打开文件，准备写入
6.  RTMP_Connect();//建立NetConnection
7.  RTMP_ConnectStream();//建立NetStream
8.  Download();//下载函数
9.  RTMP_Close();//关闭连接
10. fclose();//关闭文件
11. CleanupSockets();//清理Socket
```

其中Download()主要是使用RTMP_Read()进行下载的。

注：可以参考：[RTMP流媒体播放过程](#)

下面贴上自己注释的RTMPDump源代码。注意以下几点：

- 1.此RTMPDump已经被移植进VC 2010 的 MFC的工程,所以main()函数已经被改名为rtmpdump(),而且参数也改了,传进来一个MFC窗口的句柄。不过功能没怎么改(控制台程序移植到MFC以后,main()就不是程序的入口了,所以main()名字改成什么是无所谓的)
- 2.里面有很多提取信息的代码形如：rtmp.dlg->AppendCInfo("开始初始化Socket...");这些代码是我为了获取RTMP信息而自己加的，并不影响程序的执行。

```
[cpp]
1.  int rtmpdump(LPVOID lpParam,int argc,char **argv)
2.  {
3.
4.      extern char *optarg;
5.      //一定要设置,否则只能运行一次
6.      extern int optind;
7.      optind=0;
8.      int nStatus = RD_SUCCESS;
9.      double percent = 0;
10.     double duration = 0.0;
11.
12.     int nSkipKeyFrames = DEF_SKIPFRM; // skip this number of keyframes when resuming
13.
14.     int bOverrideBufferTime = FALSE; // if the user specifies a buffer time override this is true
15.     int bStdoutMode = TRUE; // if true print the stream directly to stdout, messages go to stderr
16.     int bResume = FALSE; // true in resume mode
17.     uint32_t dSeek = 0; // seek position in resume mode, 0 otherwise
18.     uint32_t bufferTime = DEF_BUFTIME;
19.
20.     // meta header and initial frame for the resume mode (they are read from the file and compared with
21.     // the stream we are trying to continue
22.     char *metaHeader = 0;
23.     uint32_t nMetaHeaderSize = 0;
24.
25.     // video keyframe for matching
26.     char *initialFrame = 0;
27.     uint32_t nInitialFrameSize = 0;
28.     int initialFrameType = 0; // tye: audio or video
29.
30.     AVal hostname = { 0, 0 };
31.     AVal playpath = { 0, 0 };
32.     AVal subscribepath = { 0, 0 };
33.     int port = -1;
34.     int protocol = RTMP_PROTOCOL_UNDEFINED;
35.     int retries = 0;
36.     int bLiveStream = FALSE; // 是直播流吗? then we can't seek/resume
37.     int bHashes = FALSE; // display byte counters not hashes by default
38.
39.     long int timeout = DEF_TIMEOUT; // timeout connection after 120 seconds
40.     uint32_t dStartOffset = 0; // 非直播流搜寻点seek position in non-live mode
41.     uint32_t dStopOffset = 0;
42.     RTMP rtmp = { 0 };
43.
44.     AVal swfUrl = { 0, 0 };
45.     AVal tcUrl = { 0, 0 };
46.     AVal pageUrl = { 0, 0 };
47.     AVal app = { 0, 0 };
48.     AVal auth = { 0, 0 };
49.     AVal swfHash = { 0, 0 };
50.     uint32_t swfSize = 0;
51.     AVal flashVer = { 0, 0 };
52.     AVal sockshost = { 0, 0 };
53.
54. #ifdef CRYPTO
55.     int swfAqe = 30; /* 30 days for SWF cache by default */
```

```

56.     int swfVfy = 0;
57.     unsigned char hash[RTMP_SWF_HASHLEN];
58. #endif
59.
60.     char *flvFile = 0;
61.
62.     signal(SIGINT, sigIntHandler);
63.     signal(SIGTERM, sigIntHandler);
64. #ifndef WIN32
65.     signal(SIGHUP, sigIntHandler);
66.     signal(SIGPIPE, sigIntHandler);
67.     signal(SIGQUIT, sigIntHandler);
68. #endif
69.
70.     RTMP_debuglevel = RTMP_LOGINFO;
71.
72.     //首先搜寻“--quiet”选项
73.     int index = 0;
74.     while (index < argc)
75.     {
76.         if (strcmp(argv[index], "--quiet") == 0
77.             || strcmp(argv[index], "-q") == 0)
78.             RTMP_debuglevel = RTMP_LOGCRIT;
79.         index++;
80.     }
81. #define RTMPDUMP_VERSION "1.0"
82.     RTMP_LogPrintf("RTMP流媒体下载 %s\n", RTMPDUMP_VERSION);
83.     RTMP_LogPrintf
84.     ("2012 雷霄骅 中国传媒大学/信息工程学院/通信与信息系统/数字电视技术\n");
85. //RTMP_LogPrintf("输入 -h 获取命令选项\n");
86.     RTMP_Init(&rtmp);
87.     //句柄-----
88.     rtmp.dlg=(CSpecialPRTMPDlg *)lpParam;
89.     //-----
90.     //-----
91.     rtmp.dlg->AppendCInfo("开始初始化Socket...");
92.     //-----
93.     if (!InitSockets())
94.     {
95.         //-----
96.         rtmp.dlg->AppendCInfo("初始化Socket失败!");
97.         //-----
98.         RTMP_Log(RTMP_LOGERROR,
99.             "Couldn't load sockets support on your platform, exiting!");
100.         return RD_FAILED;
101.     }
102.     //-----
103.     rtmp.dlg->AppendCInfo("成功初始化Socket");
104.     //-----
105.     /* sleep(30); */
106.
107.
108.
109.     int opt;
110.     /* struct option longopts[] = {
111.         {"help", 0, NULL, 'h'},
112.         {"host", 1, NULL, 'n'},
113.         {"port", 1, NULL, 'c'},
114.         {"socks", 1, NULL, 'S'},
115.         {"protocol", 1, NULL, 'l'},
116.         {"playpath", 1, NULL, 'y'},
117.         {"playlist", 0, NULL, 'Y'},
118.         {"rtmp", 1, NULL, 'r'},
119.         {"swfUrl", 1, NULL, 's'},
120.         {"tcUrl", 1, NULL, 't'},
121.         {"pageUrl", 1, NULL, 'p'},
122.         {"app", 1, NULL, 'a'},
123.         {"auth", 1, NULL, 'u'},
124.         {"conn", 1, NULL, 'C'},
125. #ifdef CRYPTO
126.         {"swfhash", 1, NULL, 'w'},
127.         {"swfsize", 1, NULL, 'x'},
128.         {"swfVfy", 1, NULL, 'W'},
129.         {"swfAge", 1, NULL, 'X'},
130. #endif
131.         {"flashVer", 1, NULL, 'f'},
132.         {"live", 0, NULL, 'v'},
133.         {"flv", 1, NULL, 'o'},
134.         {"resume", 0, NULL, 'e'},
135.         {"timeout", 1, NULL, 'm'},
136.         {"buffer", 1, NULL, 'b'},
137.         {"skip", 1, NULL, 'k'},
138.         {"subscribe", 1, NULL, 'd'},
139.         {"start", 1, NULL, 'A'},
140.         {"stop", 1, NULL, 'B'},
141.         {"token", 1, NULL, 'T'},
142.         {"hashes", 0, NULL, '#'},
143.         {"debug", 0, NULL, 'z'},
144.         {"quiet", 0, NULL, 'q'},
145.         {"verbose", 0, NULL, 'V'},
146.         {0, 0, 0, 0}

```

```

147. */
148. //分析命令行参数，注意用法。
149. //选项都是一个字母，后面有冒号的代表该选项还有相关参数
150. //一直循环直到获取所有的opt
151. while ((opt =
152.     getopt(/* long*/(argc, argv,
153.         "hVveqzr:s:t:p:a:b:f:o:u:C:n:c:l:y:Ym:k:d:A:B:T:w:x:W:X:S:#" /*,
154.             longopts, NULL*/)) != -1)
155. {
156.     //不同的选项做不同的处理
157.     switch (opt)
158.     {
159.     case 'h':
160.         usage(argv[0]);
161.         return RD_SUCCESS;
162. #ifdef CRYPTO
163.     case 'w':
164.     {
165.         int res = hex2bin(optarg, &swfHash.av_val);
166.         if (res != RTMP_SWF_HASHLEN)
167.         {
168.             swfHash.av_val = NULL;
169.             RTMP_Log(RTMP_LOGWARNING,
170.                 "Couldn't parse swf hash hex string, not hexstring or not %d bytes, ignoring!", RTMP_SWF_HASHLEN);
171.         }
172.         swfHash.av_len = RTMP_SWF_HASHLEN;
173.         break;
174.     }
175.     case 'x':
176.     {
177.         int size = atoi(optarg);
178.         if (size <= 0)
179.         {
180.             RTMP_Log(RTMP_LOGERROR, "SWF Size must be at least 1, ignoring\n");
181.         }
182.         else
183.         {
184.             swfSize = size;
185.         }
186.         break;
187.     }
188.     case 'W':
189.         STR2AVAL(swfUrl, optarg);
190.         swfVfy = 1;
191.         break;
192.     case 'X':
193.     {
194.         int num = atoi(optarg);
195.         if (num < 0)
196.         {
197.             RTMP_Log(RTMP_LOGERROR, "SWF Age must be non-negative, ignoring\n");
198.         }
199.         else
200.         {
201.             swfAge = num;
202.         }
203.     }
204.     break;
205. #endif
206.     case 'k':
207.         nSkipKeyFrames = atoi(optarg);
208.         if (nSkipKeyFrames < 0)
209.         {
210.             RTMP_Log(RTMP_LOGERROR,
211.                 "Number of keyframes skipped must be greater or equal zero, using zero!");
212.             nSkipKeyFrames = 0;
213.         }
214.         else
215.         {
216.             RTMP_Log(RTMP_LOGDEBUG, "Number of skipped key frames for resume: %d",
217.                 nSkipKeyFrames);
218.         }
219.         break;
220.     case 'b':
221.     {
222.         int32_t bt = atol(optarg);
223.         if (bt < 0)
224.         {
225.             RTMP_Log(RTMP_LOGERROR,
226.                 "Buffer time must be greater than zero, ignoring the specified value %d!",
227.                 bt);
228.         }
229.         else
230.         {
231.             bufferTime = bt;
232.             bOverrideBufferTime = TRUE;
233.         }
234.         break;
235.     }
236.     //直播流
237.     case 'v':

```

```

238.         //-----
239.         rtmp.dlg->AppendCInfo("该RTMP的URL是一个直播流");
240.         //-----
241.         bLiveStream = TRUE;    // no seeking or resuming possible!
242.         break;
243.     case 'd':
244.         STR2AVAL(subscribepath, optarg);
245.         break;
246.     case 'n':
247.         STR2AVAL(hostname, optarg);
248.         break;
249.     case 'c':
250.         port = atoi(optarg);
251.         break;
252.     case 'l':
253.         protocol = atoi(optarg);
254.         if (protocol < RTMP_PROTOCOL_RTMP || protocol > RTMP_PROTOCOL_RTMPSTS)
255.         {
256.             RTMP_Log(RTMP_LOGERROR, "Unknown protocol specified: %d", protocol);
257.             return RD_FAILED;
258.         }
259.         break;
260.     case 'y':
261.         STR2AVAL(playpath, optarg);
262.         break;
263.     case 'Y':
264.         RTMP_SetOpt(&rtmp, &av_playlist, (Aval *)&av_true);
265.         break;
266.         //路径参数-r
267.     case 'r':
268.         {
269.             AVal parsedHost, parsedApp, parsedPlaypath;
270.             unsigned int parsedPort = 0;
271.             int parsedProtocol = RTMP_PROTOCOL_UNDEFINED;
272.             //解析URL。 注optarg指向参数 (URL)
273.             RTMP_LogPrintf("RTMP URL : %s\n", optarg);
274.             //-----
275.             rtmp.dlg->AppendCInfo("解析RTMP的URL...");
276.             //-----
277.             if (!RTMP_ParseURL
278.                 (optarg, &parsedProtocol, &parsedHost, &parsedPort,
279.                  &parsedPlaypath, &parsedApp))
280.             {
281.                 //-----
282.                 rtmp.dlg->AppendCInfo("解析RTMP的URL失败!");
283.                 //-----
284.                 RTMP_Log(RTMP_LOGWARNING, "无法解析 url (%s)!",
285.                          optarg);
286.             }
287.             else
288.             {
289.                 //-----
290.                 rtmp.dlg->AppendCInfo("解析RTMP的URL成功");
291.                 //-----
292.                 //把解析出来的数据赋值
293.                 if (!hostname.av_len)
294.                     hostname = parsedHost;
295.                 if (port == -1)
296.                     port = parsedPort;
297.                 if (playpath.av_len == 0 && parsedPlaypath.av_len)
298.                 {
299.                     playpath = parsedPlaypath;
300.                 }
301.                 if (protocol == RTMP_PROTOCOL_UNDEFINED)
302.                     protocol = parsedProtocol;
303.                 if (app.av_len == 0 && parsedApp.av_len)
304.                 {
305.                     app = parsedApp;
306.                 }
307.             }
308.             break;
309.         }
310.     case 's':
311.         STR2AVAL(swfUrl, optarg);
312.         break;
313.     case 't':
314.         STR2AVAL(tcUrl, optarg);
315.         break;
316.     case 'p':
317.         STR2AVAL(pageUrl, optarg);
318.         break;
319.     case 'a':
320.         STR2AVAL(app, optarg);
321.         break;
322.     case 'f':
323.         STR2AVAL(flashVer, optarg);
324.         break;
325.         //指定输出文件
326.     case 'o':
327.         flvFile = optarg;
328.         if (strcmp(flvFile, "-"))

```

```

329.         bStdoutMode = FALSE;
330.
331.         break;
332.     case 'e':
333.         bResume = TRUE;
334.         break;
335.     case 'u':
336.         STR2AVAL(auth, optarg);
337.         break;
338.     case 'C': {
339.         AVal av;
340.         STR2AVAL(av, optarg);
341.         if (!RTMP_SetOpt(&rtmp, &av_conn, &av))
342.         {
343.             RTMP_Log(RTMP_LOGERROR, "Invalid AMF parameter: %s", optarg);
344.             return RD_FAILED;
345.         }
346.     }
347.     break;
348.     case 'm':
349.         timeout = atoi(optarg);
350.         break;
351.     case 'A':
352.         dStartOffset = (int) (atof(optarg) * 1000.0);
353.         break;
354.     case 'B':
355.         dStopOffset = (int) (atof(optarg) * 1000.0);
356.         break;
357.     case 'T': {
358.         AVal token;
359.         STR2AVAL(token, optarg);
360.         RTMP_SetOpt(&rtmp, &av_token, &token);
361.     }
362.     break;
363.     case '#':
364.         bHashes = TRUE;
365.         break;
366.     case 'q':
367.         RTMP_debuglevel = RTMP_LOGCRIT;
368.         break;
369.     case 'V':
370.         RTMP_debuglevel = RTMP_LOGDEBUG;
371.         break;
372.     case 'z':
373.         RTMP_debuglevel = RTMP_LOGALL;
374.         break;
375.     case 'S':
376.         STR2AVAL(sockshost, optarg);
377.         break;
378.     default:
379.         RTMP_LogPrintf("unknown option: %c\n", opt);
380.         usage(argv[0]);
381.         return RD_FAILED;
382.     break;
383. }
384. }
385.
386. if (!hostname.av_len)
387. {
388.     RTMP_Log(RTMP_LOGERROR,
389.         "您必须指定 主机名(hostname) (--host) 或 url (-r \"rtmp://host[:port]/playpath\") 包含 a hostname");
390.     return RD_FAILED;
391. }
392. if (playpath.av_len == 0)
393. {
394.     RTMP_Log(RTMP_LOGERROR,
395.         "您必须指定 播放路径(playpath) (--playpath) 或 url (-r \"rtmp://host[:port]/playpath\") 包含 a playpath");
396.     return RD_FAILED;
397. }
398.
399. if (protocol == RTMP_PROTOCOL_UNDEFINED)
400. {
401.     RTMP_Log(RTMP_LOGWARNING,
402.         "您没有指定 协议(protocol) (--protocol) 或 rtmp url (-r), 默认协议 RTMP");
403.     protocol = RTMP_PROTOCOL_RTMP;
404. }
405. if (port == -1)
406. {
407.     RTMP_Log(RTMP_LOGWARNING,
408.         "您没有指定 端口(port) (--port) 或 rtmp url (-r), 默认端口 1935");
409.     port = 0;
410. }
411. if (port == 0)
412. {
413.     if (protocol & RTMP_FEATURE_SSL)
414.         port = 443;
415.     else if (protocol & RTMP_FEATURE_HTTP)
416.         port = 80;
417.     else
418.         port = 1935;
419. }

```

```

420.
421.     if (flvFile == 0)
422.     {
423.         RTMP_Log(RTMP_LOGWARNING,
424.             "请指定一个输出文件 (-o filename), using stdout");
425.         bStdoutMode = TRUE;
426.     }
427.
428.     if (bStdoutMode && bResume)
429.     {
430.         RTMP_Log(RTMP_LOGWARNING,
431.             "Can't resume in stdout mode, ignoring --resume option");
432.         bResume = FALSE;
433.     }
434.
435.     if (bLiveStream && bResume)
436.     {
437.         RTMP_Log(RTMP_LOGWARNING, "Can't resume live stream, ignoring --resume option");
438.         bResume = FALSE;
439.     }
440.
441. #ifdef CRYPTO
442.     if (swfVfy)
443.     {
444.         if (RTMP_HashSWF(swfUrl.av_val, (unsigned int *)&swfSize, hash, swfAge) == 0)
445.         {
446.             swfHash.av_val = (char *)hash;
447.             swfHash.av_len = RTMP_SWF_HASHLEN;
448.         }
449.     }
450.
451.     if (swfHash.av_len == 0 && swfSize > 0)
452.     {
453.         RTMP_Log(RTMP_LOGWARNING,
454.             "Ignoring SWF size, supply also the hash with --swfhash");
455.         swfSize = 0;
456.     }
457.
458.     if (swfHash.av_len != 0 && swfSize == 0)
459.     {
460.         RTMP_Log(RTMP_LOGWARNING,
461.             "Ignoring SWF hash, supply also the swf size with --swfsize");
462.         swfHash.av_len = 0;
463.         swfHash.av_val = NULL;
464.     }
465. #endif
466.
467.     if (tcUrl.av_len == 0)
468.     {
469.         char str[512] = { 0 };
470.
471.         tcUrl.av_len = snprintf(str, 511, "%s://%s:%d/%s",
472.             RTMPProtocolStringsLower[protocol], hostname.av_val,
473.             port, app.av_val, app.av_val);
474.         tcUrl.av_val = (char *) malloc(tcUrl.av_len + 1);
475.         strcpy(tcUrl.av_val, str);
476.     }
477.
478.     int first = 1;
479.
480.     // User defined seek offset
481.     if (dStartOffset > 0)
482.     {
483.         //直播流
484.         if (bLiveStream)
485.         {
486.             RTMP_Log(RTMP_LOGWARNING,
487.                 "Can't seek in a live stream, ignoring --start option");
488.             dStartOffset = 0;
489.         }
490.     }
491.     //-----
492.     rtmp.dlg->AppendCInfo("开始初始化RTMP连接的参数...");
493.     //-----
494.     //设置
495.     RTMP_SetupStream(&rtmp, protocol, &hostname, port, &sockshost, &playpath,
496.         &tcUrl, &swfUrl, &pageUrl, &app, &auth, &swfHash, swfSize,
497.         &flashVer, &subscribepath, dSeek, dStopOffset, bLiveStream, timeout);
498.     //此处设置参数-----
499.     rtmp.dlg->AppendCInfo("成功初始化RTMP连接的参数");
500.     //-----
501.     char *temp=(char *)malloc(MAX_URL_LENGTH);
502.
503.     memcpy(temp,rtmp.Link.hostname.av_val,rtmp.Link.hostname.av_len);
504.     temp[rtmp.Link.hostname.av_len]='\0';
505.     rtmp.dlg->AppendB_R_L_Info("主机名",temp);
506.
507.     itoa(rtmp.Link.port,temp,10);
508.     rtmp.dlg->AppendB_R_L_Info("端口号",temp);
509.
510.     memcpy(temp,rtmp.Link.app.av_val,rtmp.Link.app.av_len);
511.     temp[rtmp.Link.app.av_len]='\0';

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512.     rtmp.dlg->AppendB_L_Info("应用程序",temp);
513.
514.     memcpy(temp,rtmp.Link.playpath.av_val,rtmp.Link.playpath.av_len);
515.     temp[rtmp.Link.playpath.av_len]='\0';
516.     rtmp.dlg->AppendB_R_L_Info("路径",temp);
517.
518.
519.     //-----
520.
521.     /* Try to keep the stream moving if it pauses on us */
522.     if (!bLiveStream && !(protocol & RTMP_FEATURE_HTTP))
523.         rtmp.Link.lFlags |= RTMP_LF_BUFEX;
524.
525.     off_t size = 0;
526.
527.     // ok,我们必须获得timestamp of the last keyframe (only keyframes are seekable) / last audio frame (audio only streams)
528.     if (bResume)
529.     {
530.         //打开文件, 输出的文件(Resume)
531.         nStatus =
532.         OpenResumeFile(flvFile, &file, &size, &metaHeader, &nMetaHeaderSize,
533.             &duration);
534.         if (nStatus == RD_FAILED)
535.             goto clean;
536.
537.         if (!file)
538.         {
539.             // file does not exist, so go back into normal mode
540.             bResume = FALSE; // we are back in fresh file mode (otherwise finalizing file won't be done)
541.         }
542.         else
543.         {
544.             //获取最后一个关键帧
545.             nStatus = GetLastKeyframe(file, nSkipKeyFrames,
546.                 &dSeek, &initialFrame,
547.                 &initialFrameType, &nInitialFrameSize);
548.             if (nStatus == RD_FAILED)
549.             {
550.                 RTMP_Log(RTMP_LOGDEBUG, "Failed to get last keyframe.");
551.                 goto clean;
552.             }
553.
554.             if (dSeek == 0)
555.             {
556.                 RTMP_Log(RTMP_LOGDEBUG,
557.                     "Last keyframe is first frame in stream, switching from resume to normal mode!");
558.                 bResume = FALSE;
559.             }
560.         }
561.     }
562.     //如果输出文件不存在
563.     if (!file)
564.     {
565.         if (bStdoutMode)
566.         {
567.             //直接输出到stdout
568.             file = stdout;
569.             SET_BINMODE(file);
570.         }
571.         else
572.         {
573.             //打开一个文件
574.             //w+b 读写打开或建立一个二进制文件, 允许读和写。
575.             //-----
576.             rtmp.dlg->AppendCInfo("创建输出文件...");
577.             //-----
578.             file = fopen(flvFile, "w+b");
579.             if (file == 0)
580.             {
581.                 //-----
582.                 rtmp.dlg->AppendCInfo("创建输出文件失败!");
583.                 //-----
584.                 RTMP_LogPrintf("Failed to open file! %s\n", flvFile);
585.                 return RD_FAILED;
586.             }
587.             rtmp.dlg->AppendCInfo("成功创建输出文件");
588.         }
589.     }
590.
591. #ifdef _DEBUG
592.     netstackdump = fopen("netstackdump", "wb");
593.     netstackdump_read = fopen("netstackdump_read", "wb");
594. #endif
595.
596.     while (!RTMP_ctrlC)
597.     {
598.         RTMP_Log(RTMP_LOGDEBUG, "Setting buffer time to: %dms", bufferTime);
599.         //设置Buffer时间
600.         //-----
601.         rtmp.dlg->AppendCInfo("设置缓冲(Buffer)的时间");
602.         //-----

```



```

603.     RTMP_SetBufferMS(&rtmp, bufferTime);
604.     //第一次执行
605.     if (first)
606.     {
607.         first = 0;
608.         RTMP_LogPrintf("开始建立连接!\n");
609.         //-----
610.         rtmp.dlg->AppendCInfo("开始建立连接 (NetConnection) ...");
611.         //-----
612.         //建立连接(Connect)
613.         if (!RTMP_Connect(&rtmp, NULL))
614.         {
615.             //-----
616.             rtmp.dlg->AppendCInfo("建立连接 (NetConnection) 失败!");
617.             //-----
618.             nStatus = RD_FAILED;
619.             break;
620.         }
621.         //-----
622.         rtmp.dlg->AppendCInfo("成功建立连接 (NetConnection) ");
623.         //-----
624.         //RTMP_Log(RTMP_LOGINFO, "已链接...");
625.
626.         // User defined seek offset
627.         if (dStartOffset > 0)
628.         {
629.             // Don't need the start offset if resuming an existing file
630.             if (bResume)
631.             {
632.                 RTMP_Log(RTMP_LOGWARNING,
633.                     "Can't seek a resumed stream, ignoring --start option");
634.                 dStartOffset = 0;
635.             }
636.             else
637.             {
638.                 dSeek = dStartOffset;
639.             }
640.         }
641.
642.         // Calculate the length of the stream to still play
643.         if (dStopOffset > 0)
644.         {
645.             // Quit if start seek is past required stop offset
646.             if (dStopOffset <= dSeek)
647.             {
648.                 RTMP_LogPrintf("Already Completed\n");
649.                 nStatus = RD_SUCCESS;
650.                 break;
651.             }
652.         }
653.         //创建流(Stream) (发送connect命令消息后处理传来的数据)
654.         itoa(rtmp.m_inChunkSize, temp, 10);
655.         rtmp.dlg->AppendB_R_Info("输入Chunk大小", temp);
656.         itoa(rtmp.m_outChunkSize, temp, 10);
657.         rtmp.dlg->AppendB_R_Info("输出Chunk大小", temp);
658.         itoa(rtmp.m_stream_id, temp, 10);
659.         rtmp.dlg->AppendB_R_Info("Stream ID", temp);
660.         itoa(rtmp.m_nBufferMS, temp, 10);
661.         rtmp.dlg->AppendB_R_Info("Buffer时长 (ms) ", temp);
662.         itoa(rtmp.m_nServerBW, temp, 10);
663.         rtmp.dlg->AppendB_R_Info("ServerBW", temp);
664.         itoa(rtmp.m_nClientBW, temp, 10);
665.         rtmp.dlg->AppendB_R_Info("ClientBW", temp);
666.         itoa((int)rtmp.m_fEncoding, temp, 10);
667.         rtmp.dlg->AppendB_R_Info("命令消息编码方法", temp);
668.         itoa((int)rtmp.m_fDuration, temp, 10);
669.         rtmp.dlg->AppendB_R_Info("时长 (s) ", temp);
670.
671.         rtmp.dlg->ShowBInfo();
672.         free(temp);
673.         //-----
674.         rtmp.dlg->AppendCInfo("开始建立网络流 (NetStream) ");
675.         //-----
676.         if (!RTMP_ConnectStream(&rtmp, dSeek))
677.         {
678.             //-----
679.             rtmp.dlg->AppendCInfo("建立网络流 (NetStream) 失败!");
680.             //-----
681.             nStatus = RD_FAILED;
682.             break;
683.         }
684.         //-----
685.         rtmp.dlg->AppendCInfo("成功建立网络流 (NetStream) !");
686.         //-----
687.     }
688.     else
689.     {
690.         nInitialFrameSize = 0;
691.
692.         if (retries)
693.         {

```

```

694.     RTMP_Log(RTMP_LOGERROR, "Failed to resume the stream\n\n");
695.     if (!RTMP_IsTimedout(&rtmp))
696.         nStatus = RD_FAILED;
697.     else
698.         nStatus = RD_INCOMPLETE;
699.     break;
700. }
701. RTMP_Log(RTMP_LOGINFO, "Connection timed out, trying to resume.\n\n");
702. /* Did we already try pausing, and it still didn't work? */
703. if (rtmp.m_pausing == 3)
704. {
705.     /* Only one try at reconnecting... */
706.     retries = 1;
707.     dSeek = rtmp.m_pauseStamp;
708.     if (dStopOffset > 0)
709.     {
710.         if (dStopOffset <= dSeek)
711.         {
712.             RTMP_LogPrintf("Already Completed\n");
713.             nStatus = RD_SUCCESS;
714.             break;
715.         }
716.     }
717.     if (!RTMP_ReconnectStream(&rtmp, dSeek))
718.     {
719.         RTMP_Log(RTMP_LOGERROR, "Failed to resume the stream\n\n");
720.         if (!RTMP_IsTimedout(&rtmp))
721.             nStatus = RD_FAILED;
722.         else
723.             nStatus = RD_INCOMPLETE;
724.         break;
725.     }
726. }
727. else if (!RTMP_ToggleStream(&rtmp))
728. {
729.     RTMP_Log(RTMP_LOGERROR, "Failed to resume the stream\n\n");
730.     if (!RTMP_IsTimedout(&rtmp))
731.         nStatus = RD_FAILED;
732.     else
733.         nStatus = RD_INCOMPLETE;
734.     break;
735. }
736. bResume = TRUE;
737. }
738. //-----
739.
740. //-----
741. rtmp.dlg->AppendCInfo("开始将媒体数据写入文件");
742. //-----
743. //下载,写入文件
744. nStatus = Download(&rtmp, file, dSeek, dStopOffset, duration, bResume,
745.     metaHeader, nMetaHeaderSize, initialFrame,
746.     initialFrameType, nInitialFrameSize,
747.     nSkipKeyFrames, bStdoutMode, bLiveStream, bHashes,
748.     bOverrideBufferTime, bufferTime, &percent);
749. free(initialFrame);
750. initialFrame = NULL;
751.
752. /* If we succeeded, we're done.
753. */
754. if (nStatus != RD_INCOMPLETE || !RTMP_IsTimedout(&rtmp) || bLiveStream)
755. break;
756. }
757. //下载完的时候
758. if (nStatus == RD_SUCCESS)
759. {
760.     //-----
761.     rtmp.dlg->AppendCInfo("写入文件完成");
762.     //-----
763.     RTMP_LogPrintf("Download complete\n");
764. }
765. //没下载完的时候
766. else if (nStatus == RD_INCOMPLETE)
767. {
768.     //-----
769.     rtmp.dlg->AppendCInfo("写入文件可能不完整");
770.     //-----
771.     RTMP_LogPrintf
772. ("Download may be incomplete (downloaded about %.2f%%), try resuming\n",
773.     percent);
774. }
775. //后续清理工作
776. clean:
777. //-----
778. rtmp.dlg->AppendCInfo("关闭连接");
779. //-----
780. RTMP_Log(RTMP_LOGDEBUG, "Closing connection.\n");
781. RTMP_Close(&rtmp);
782. rtmp.dlg->AppendCInfo("关闭文件");
783. if (file != 0)
784.     fclose(file);

```

```

785.     rtmp.dlg->AppendCInfo("关闭Socket");
786.     CleanupSockets();
787.
788. #ifdef _DEBUG
789.     if (netstackdump != 0)
790.         fclose(netstackdump);
791.     if (netstackdump_read != 0)
792.         fclose(netstackdump_read);
793. #endif
794.     return nStatus;
795. }

```

其中InitSocket()代码很简单，初始化了Socket，如下：

```

1. // 初始化 sockets
2. int
3. InitSockets()
4. {
5. #ifdef WIN32
6.     WORD version;
7.     WSADATA wsaData;
8.
9.     version = MAKEWORD(1, 1);
10.    return (WSAStartup(version, &wsaData) == 0);
11. #else
12.    return TRUE;
13. #endif
14. }

```

CleanupSockets()则更简单：

```

1. inline void
2. CleanupSockets()
3. {
4. #ifdef WIN32
5.     WSACleanup();
6. #endif
7. }

```

Download()函数则比较复杂：

```

1. int
2. Download(RTMP * rtmp, // connected RTMP object
3.     FILE * file, uint32_t dSeek, uint32_t dStopOffset, double duration, int bResume, char *metaHeader, uint32_t nMetaHeaderSize, char *initialFrame, int initialFrameType, uint32_t nInitialFrameSize, int nSkipKeyFrames, int bStdoutMode, int bLiveStream, int bHashes, int bOverrideBufferTime, uint32_t bufferTime, double *percent) // percentage downloaded [out]
4. {
5.     int32_t now, lastUpdate;
6.     int bufferSize = 64 * 1024;
7.     char *buffer = (char *) malloc(bufferSize);
8.     int nRead = 0;
9.
10.    //long ftell(FILE *stream);
11.    //返回当前文件指针
12.    RTMP_LogPrintf("开始下载！\n");
13.    off_t size = ftello(file);
14.    unsigned long lastPercent = 0;
15.    //时间戳
16.    rtmp->m_read.timestamp = dSeek;
17.
18.    *percent = 0.0;
19.
20.    if (rtmp->m_read.timestamp)
21.    {
22.        RTMP_Log(RTMP_LOGDEBUG, "Continuing at TS: %d ms\n", rtmp->m_read.timestamp);
23.    }
24.    //是直播
25.    if (bLiveStream)
26.    {
27.        RTMP_LogPrintf("直播流\n");
28.    }
29.    else
30.    {
31.        // print initial status
32.        // Workaround to exit with 0 if the file is fully (> 99.9%) downloaded
33.        if (duration > 0)
34.        {
35.            if ((double) rtmp->m_read.timestamp >= (double) duration * 999.0)
36.            {
37.                RTMP_LogPrintf("Already Completed at: %.3f sec Duration=%.3f sec\n",
38.                    (double) rtmp->m_read.timestamp / 1000.0,
39.                    (double) duration / 1000.0);

```

```

40.         return RD_SUCCESS;
41.     }
42.     else
43.     {
44.         *percent = ((double) rtmp->m_read.timestamp) / (duration * 1000.0) * 100.0;
45.         *percent = ((double) (int) (*percent * 10.0)) / 10.0;
46.         RTMP_LogPrintf("%s download at: %.3f kB / %.3f sec (%.1f%%)\n",
47.             bResume ? "Resuming" : "Starting",
48.             (double) size / 1024.0, (double) rtmp->m_read.timestamp / 1000.0,
49.             *percent);
50.     }
51. }
52. else
53. {
54.     RTMP_LogPrintf("%s download at: %.3f kB\n",
55.         bResume ? "Resuming" : "Starting",
56.         (double) size / 1024.0);
57. }
58. }
59.
60. if (dStopOffset > 0)
61.     RTMP_LogPrintf("For duration: %.3f sec\n", (double) (dStopOffset - dSeek) / 1000.0);
62.
63. //各种设置参数到rtmp连接
64. if (bResume && nInitialFrameSize > 0)
65.     rtmp->m_read.flags |= RTMP_READ_RESUME;
66. rtmp->m_read.initialFrameType = initialFrameType;
67. rtmp->m_read.nResumeTS = dSeek;
68. rtmp->m_read.metaHeader = metaHeader;
69. rtmp->m_read.initialFrame = initialFrame;
70. rtmp->m_read.nMetaHeaderSize = nMetaHeaderSize;
71. rtmp->m_read.nInitialFrameSize = nInitialFrameSize;
72.
73. now = RTMP_GetTime();
74. lastUpdate = now - 1000;
75. do
76. {
77.     //从rtmp中把bufferSize (64k) 个数据读入buffer
78.     nRead = RTMP_Read(rtmp, buffer, bufferSize);
79.     //RTMP_LogPrintf("nRead: %d\n", nRead);
80.     if (nRead > 0)
81.     {
82.         //函数: size_t fwrite(const void* buffer, size_t size, size_t count, FILE* stream);
83.         //向文件读入写入一个数据块。返回值: 返回实际写入的数据块数目
84.         // (1) buffer: 是一个指针, 对fwrite来说, 是要输出数据的地址。
85.         // (2) size: 要写入内容的单字节数;
86.         // (3) count: 要进行写入size字节的数据项的个数;
87.         // (4) stream: 目标文件指针。
88.         // (5) 返回实际写入的数据项个数count。
89.         //关键: 把buffer里面的数据写成文件
90.         if (fwrite(buffer, sizeof(unsigned char), nRead, file) !=
91.             (size_t) nRead)
92.         {
93.             RTMP_Log(RTMP_LOGERROR, "%s: Failed writing, exiting!", __FUNCTION__);
94.             free(buffer);
95.             return RD_FAILED;
96.         }
97.         //记录已经写入的字节数
98.         size += nRead;
99.
100.        //RTMP_LogPrintf("write %dbytes (%.1f kB)\n", nRead, nRead/1024.0);
101.        if (duration <= 0) // if duration unknown try to get it from the stream (onMetaData)
102.            duration = RTMP_GetDuration(rtmp);
103.
104.        if (duration > 0)
105.        {
106.            // make sure we claim to have enough buffer time!
107.            if (!bOverrideBufferTime && bufferTime < (duration * 1000.0))
108.            {
109.                bufferTime = (uint32_t) (duration * 1000.0) + 5000;    // 再加5s以确保buffertime足够长
110.
111.                RTMP_Log(RTMP_LOGDEBUG,
112.                    "Detected that buffer time is less than duration, resetting to: %dms",
113.                    bufferTime);
114.                //重设Buffer长度
115.                RTMP_SetBufferMS(rtmp, bufferTime);
116.                //给服务器发送UserControl消息通知Buffer改变
117.                RTMP_UpdateBufferMS(rtmp);
118.            }
119.
120.            //计算百分比
121.            *percent = ((double) rtmp->m_read.timestamp) / (duration * 1000.0) * 100.0;
122.            *percent = ((double) (int) (*percent * 10.0)) / 10.0;
123.            if (bHashes)
124.            {
125.                if (lastPercent + 1 <= *percent)
126.                {
127.                    RTMP_LogStatus("#");
128.                    lastPercent = (unsigned long) *percent;
129.                }
130.            }
131.        }
132.    }
133. }

```

```

131.     {
132.         //设置显示数据的更新间隔200ms
133.         now = RTMP_GetTime();
134.         if (abs(now - lastUpdate) > 200)
135.         {
136.             RTMP_LogStatus("\r%.3f kB / %.2f sec (%.1f%%)",
137.                 (double) size / 1024.0,
138.                 (double) (rtmp->m_read.timestamp) / 1000.0, *percent);
139.             lastUpdate = now;
140.         }
141.     }
142. }
143. else
144. {
145.     //现在距离开机的毫秒数
146.     now = RTMP_GetTime();
147.     //每隔200ms刷新一次数据
148.     if (abs(now - lastUpdate) > 200)
149.     {
150.         if (bHashes)
151.             RTMP_LogStatus("#");
152.         else
153.             //size为已写入文件的字节数
154.             RTMP_LogStatus("\r%.3f kB / %.2f sec", (double) size / 1024.0,
155.                 (double) (rtmp->m_read.timestamp) / 1000.0);
156.         lastUpdate = now;
157.     }
158. }
159. }
160. #ifdef _DEBUG
161.     else
162.     {
163.         RTMP_Log(RTMP_LOGDEBUG, "zero read!");
164.     }
165. #endif
166.
167. }
168. while (!RTMP_ctrlC && nRead > -1 && RTMP_IsConnected(rtmp) && !RTMP_IsTimedout(rtmp));
169. free(buffer);
170. if (nRead < 0)
171.     //nRead是读取情况
172.     nRead = rtmp->m_read.status;
173.
174. /* Final status update */
175. if (!bHashes)
176. {
177.     if (duration > 0)
178.     {
179.         *percent = ((double) rtmp->m_read.timestamp) / (duration * 1000.0) * 100.0;
180.         *percent = ((double) (int) (*percent * 10.0)) / 10.0;
181.         //输出
182.         RTMP_LogStatus("\r%.3f kB / %.2f sec (%.1f%%)",
183.             (double) size / 1024.0,
184.             (double) (rtmp->m_read.timestamp) / 1000.0, *percent);
185.     }
186.     else
187.     {
188.         RTMP_LogStatus("\r%.3f kB / %.2f sec", (double) size / 1024.0,
189.             (double) (rtmp->m_read.timestamp) / 1000.0);
190.     }
191. }
192.
193. RTMP_Log(RTMP_LOGDEBUG, "RTMP_Read returned: %d", nRead);
194. //读取错误
195. if (bResume && nRead == -2)
196. {
197.     RTMP_LogPrintf("Couldn't resume FLV file, try --skip %d\n\n",
198.         nSkipKeyFrames + 1);
199.     return RD_FAILED;
200. }
201. //读取正确
202. if (nRead == -3)
203.     return RD_SUCCESS;
204. //没读完...
205. if ((duration > 0 && *percent < 99.9) || RTMP_ctrlC || nRead < 0
206.     || RTMP_IsTimedout(rtmp))
207. {
208.     return RD_INCOMPLETE;
209. }
210.
211. return RD_SUCCESS;
212. }

```

以上内容是我能理解到的rtmpdump.c里面的内容。

rtmpdump源代码（VC 2005 工程）：<http://download.csdn.net/detail/leixiaohua1020/6563163>

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