原 RTMPdump 源代码分析 1： main()函数

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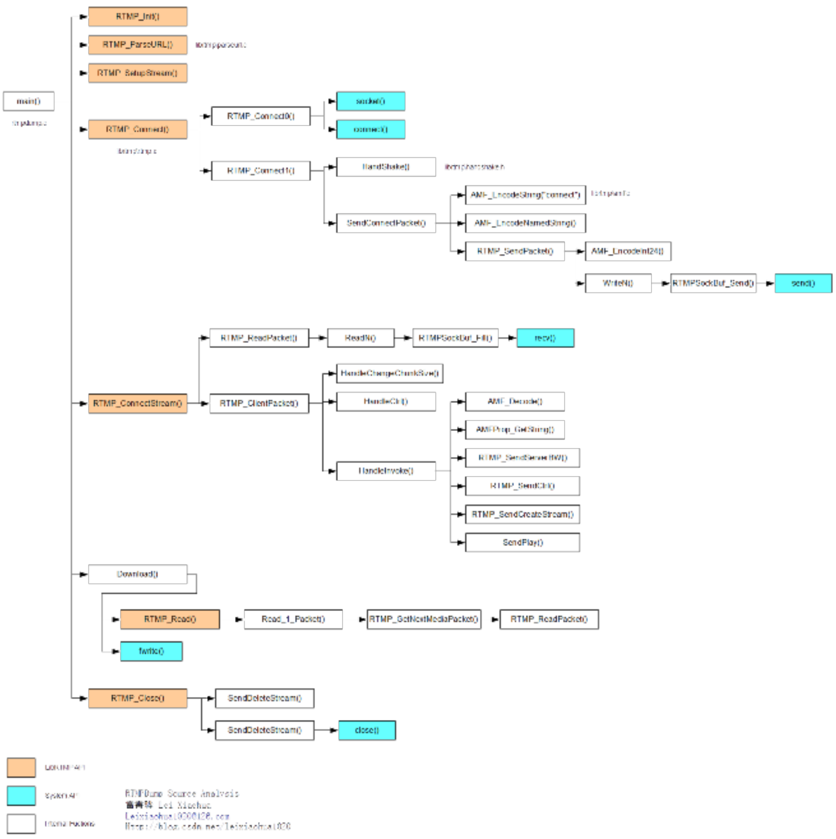
rtmpdump 是一个用来处理 RTMP 流媒体的工具包，支持 rtmp://, rtmpt://, rtmpe://, rtmpte://, and rtmps:// 等。之前在学习RTMP协议的时候，发现没有讲它源代码的，只好自己分析，现在打算把自己学习的成果写出来，可能结果不一定都对，先暂且记录一下。

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]

```cpp
RTMP_Init();//初始化结构体
InitSockets();//初始化Socket
RTMP_ParseURL();//解析输入URL
RTMP_SetupStream();//一些设置
fopen();//打开文件，准备写入
RTMP_Connect();//建立NetConnection
RTMP_ConnectStream()//建立NetStream
Download();//下载函数
RTMP_Close();//关闭连接
fclose();//关闭文件
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]

```cpp
int rtmpdump(LPVOID lpParam,int argc,char **argv)
{

  extern char *optarg;
  //一定要设置，否则只能运行一次
  extern int optind;
  optind=0;
  int nStatus = RD_SUCCESS;
  double percent = 0;
  double duration = 0.0;

  int nSkipKeyFrames = DEF_SKIPFRM; // skip this number of keyframes when resuming

  int bOverrideBufferTime = FALSE;  // if the user specifies a buffer time override this is true
  int bStdoutMode = TRUE;   // if true print the stream directly to stdout, messages go to stderr
  int bResume = FALSE;      // true in resume mode
  uint32_t dSeek = 0;       // seek position in resume mode, 0 otherwise
  uint32_t bufferTime = DEF_BUFTIME;

  // meta header and initial frame for the resume mode (they are read from the file and compared with
  // the stream we are trying to continue
  char *metaHeader = 0;
  uint32_t nMetaHeaderSize = 0;

  // video keyframe for matching
  char *initialFrame = 0;
  uint32_t nInitialFrameSize = 0;
  int initialFrameType = 0; // tye: audio or video

  AVal hostname = { 0, 0 };
  AVal playpath = { 0, 0 };
  AVal subscribepath = { 0, 0 };
  int port = -1;
  int protocol = RTMP_PROTOCOL_UNDEFINED;
  int retries = 0;
  int bLiveStream = FALSE;  // 是直播流吗? then we can't seek/resume
  int bHashes = FALSE;      // display byte counters not hashes by default

  long int timeout = DEF_TIMEOUT;   // timeout connection after 120 seconds
  uint32_t dStartOffset = 0;    // 非直播流搜寻点seek position in non-live mode
  uint32_t dStopOffset = 0;
  RTMP rtmp = { 0 };

  AVal swfUrl = { 0, 0 };
  AVal tcUrl = { 0, 0 };
  AVal pageUrl = { 0, 0 };
  AVal app = { 0, 0 };
  AVal auth = { 0, 0 };
  AVal swfHash = { 0, 0 };
  uint32_t swfSize = 0;
  AVal flashVer = { 0, 0 };
  AVal sockshost = { 0, 0 };

#ifdef CRYPTO
  int swfAge = 30;  /* 30 days for SWF cache by default */
```

```
  int swfVfy = 0;
  unsigned char hash[RTMP_SWF_HASHLEN];
#endif

  char *flvFile = 0;

  signal(SIGINT, sigIntHandler);
  signal(SIGTERM, sigIntHandler);
#ifndef WIN32
  signal(SIGHUP, sigIntHandler);
  signal(SIGPIPE, sigIntHandler);
  signal(SIGQUIT, sigIntHandler);
#endif

  RTMP_debuglevel = RTMP_LOGINFO;

  //首先搜寻“ --quiet”选项
  int index = 0;
  while (index < argc)
    {
      if (strcmp(argv[index], "--quiet") == 0
      || strcmp(argv[index], "-q") == 0)
    RTMP_debuglevel = RTMP_LOGCRIT;
      index++;
    }
#define RTMPDUMP_VERSION "1.0"
  RTMP_LogPrintf("RTMP流媒体下载 %s\n", RTMPDUMP_VERSION);
  RTMP_LogPrintf
    ("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");
  //RTMP_LogPrintf("输入 -h 获取命令选项\n");
    RTMP_Init(&rtmp);
    //句柄---------------------------
    rtmp.dlg=(CSpecialPRTMPDlg *)lpParam;
    //-------------------------------
    //---------------------
    rtmp.dlg->AppendCInfo("开始初始化Socket...");
    //---------------------------
    if (!InitSockets())
    {
        //---------------------
        rtmp.dlg->AppendCInfo("初始化Socket失败！");
        //---------------------------
        RTMP_Log(RTMP_LOGERROR,
            "Couldn't load sockets support on your platform, exiting!");
        return RD_FAILED;
    }
  //---------------------
  rtmp.dlg->AppendCInfo("成功初始化Socket");
  //---------------------------
  /* sleep(30); */



  int opt;
/*  struct option longopts[] = {
    {"help", 0, NULL, 'h'},
    {"host", 1, NULL, 'n'},
    {"port", 1, NULL, 'c'},
    {"socks", 1, NULL, 'S'},
    {"protocol", 1, NULL, 'l'},
    {"playpath", 1, NULL, 'y'},
    {"playlist", 0, NULL, 'Y'},
    {"rtmp", 1, NULL, 'r'},
    {"swfUrl", 1, NULL, 's'},
    {"tcUrl", 1, NULL, 't'},
    {"pageUrl", 1, NULL, 'p'},
    {"app", 1, NULL, 'a'},
    {"auth", 1, NULL, 'u'},
    {"conn", 1, NULL, 'C'},
#ifdef CRYPTO
    {"swfhash", 1, NULL, 'w'},
    {"swfsize", 1, NULL, 'x'},
    {"swfVfy", 1, NULL, 'W'},
    {"swfAge", 1, NULL, 'X'},
#endif
    {"flashVer", 1, NULL, 'f'},
    {"live", 0, NULL, 'v'},
    {"flv", 1, NULL, 'o'},
    {"resume", 0, NULL, 'e'},
    {"timeout", 1, NULL, 'm'},
    {"buffer", 1, NULL, 'b'},
    {"skip", 1, NULL, 'k'},
    {"subscribe", 1, NULL, 'd'},
    {"start", 1, NULL, 'A'},
    {"stop", 1, NULL, 'B'},
    {"token", 1, NULL, 'T'},
    {"hashes", 0, NULL, '#'},
    {"debug", 0, NULL, 'z'},
    {"quiet", 0, NULL, 'q'},
    {"verbose", 0, NULL, 'V'},
    {0, 0, 0, 0}
```

```
  };*/
  //分析命令行参数，注意用法。
  //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
  //一直循环直到获取所有的opt
  while ((opt =
      getopt/*_long*/(argc, argv,
              "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:#"/*,
              longopts, NULL*/)) != -1)
    {
    //不同的选项做不同的处理
      switch (opt)
    {
    case 'h':
      usage(argv[0]);
      return RD_SUCCESS;
#ifdef CRYPTO
    case 'w':
      {
        int res = hex2bin(optarg, &swfHash.av_val);
        if (res != RTMP_SWF_HASHLEN)
          {
        swfHash.av_val = NULL;
        RTMP_Log(RTMP_LOGWARNING,
            "Couldn't parse swf hash hex string, not hexstring or not %d bytes, ignoring!", RTMP_SWF_HASHLEN);
          }
        swfHash.av_len = RTMP_SWF_HASHLEN;
        break;
      }
    case 'x':
      {
        int size = atoi(optarg);
        if (size <= 0)
          {
        RTMP_Log(RTMP_LOGERROR, "SWF Size must be at least 1, ignoring\n");
          }
        else
          {
        swfSize = size;
          }
        break;
      }
        case 'W':
      STR2AVAL(swfUrl, optarg);
      swfVfy = 1;
          break;
        case 'X':
      {
        int num = atoi(optarg);
        if (num < 0)
          {
        RTMP_Log(RTMP_LOGERROR, "SWF Age must be non-negative, ignoring\n");
          }
        else
          {
        swfAge = num;
          }
      }
          break;
#endif
    case 'k':
      nSkipKeyFrames = atoi(optarg);
      if (nSkipKeyFrames < 0)
        {
          RTMP_Log(RTMP_LOGERROR,
          "Number of keyframes skipped must be greater or equal zero, using zero!");
          nSkipKeyFrames = 0;
        }
      else
        {
          RTMP_Log(RTMP_LOGDEBUG, "Number of skipped key frames for resume: %d",
          nSkipKeyFrames);
        }
      break;
    case 'b':
      {
        int32_t bt = atol(optarg);
        if (bt < 0)
          {
        RTMP_Log(RTMP_LOGERROR,
            "Buffer time must be greater than zero, ignoring the specified value %d!",
            bt);
          }
        else
          {
        bufferTime = bt;
        bOverrideBufferTime = TRUE;
          }
        break;
      }
    //直播流
    case 'v':
```

```
        //----------------
        rtmp.dlg->AppendCInfo("该RTMP的URL是一个直播流");
        //----------------
      bLiveStream = TRUE;   // no seeking or resuming possible!
      break;
    case 'd':
      STR2AVAL(subscribepath, optarg);
      break;
    case 'n':
      STR2AVAL(hostname, optarg);
      break;
    case 'c':
      port = atoi(optarg);
      break;
    case 'l':
      protocol = atoi(optarg);
      if (protocol < RTMP_PROTOCOL_RTMP || protocol > RTMP_PROTOCOL_RTMPTS)
        {
          RTMP_Log(RTMP_LOGERROR, "Unknown protocol specified: %d", protocol);
          return RD_FAILED;
        }
      break;
    case 'y':
      STR2AVAL(playpath, optarg);
      break;
    case 'Y':
      RTMP_SetOpt(&rtmp, &av_playlist, (AVal *)&av_true);
      break;
      //路径参数-r
    case 'r':
      {
        AVal parsedHost, parsedApp, parsedPlaypath;
        unsigned int parsedPort = 0;
        int parsedProtocol = RTMP_PROTOCOL_UNDEFINED;
        //解析URL。注optarg指向参数（URL）
        RTMP_LogPrintf("RTMP URL : %s\n",optarg);
        //----------------
        rtmp.dlg->AppendCInfo("解析RTMP的URL...");
        //----------------
        if (!RTMP_ParseURL
        (optarg, &parsedProtocol, &parsedHost, &parsedPort,
         &parsedPlaypath, &parsedApp))
          {
            //----------------
            rtmp.dlg->AppendCInfo("解析RTMP的URL失败！");
            //----------------
        RTMP_Log(RTMP_LOGWARNING, "无法解析 url (%s)!",
            optarg);
          }
        else
          {
            //----------------
            rtmp.dlg->AppendCInfo("解析RTMP的URL成功");
            //----------------
        //把解析出来的数据赋值
        if (!hostname.av_len)
          hostname = parsedHost;
        if (port == -1)
          port = parsedPort;
        if (playpath.av_len == 0 && parsedPlaypath.av_len)
          {
            playpath = parsedPlaypath;
          }
        if (protocol == RTMP_PROTOCOL_UNDEFINED)
          protocol = parsedProtocol;
        if (app.av_len == 0 && parsedApp.av_len)
          {
            app = parsedApp;
          }
          }
          break;
      }
    case 's':
      STR2AVAL(swfUrl, optarg);
      break;
    case 't':
      STR2AVAL(tcUrl, optarg);
      break;
    case 'p':
      STR2AVAL(pageUrl, optarg);
      break;
    case 'a':
      STR2AVAL(app, optarg);
      break;
    case 'f':
      STR2AVAL(flashVer, optarg);
      break;
    //指定输出文件
    case 'o':
      flvFile = optarg;
      if (strcmp(flvFile, "-"))
```

```
        bStdoutMode = FALSE;

      break;
    case 'e':
      bResume = TRUE;
      break;
    case 'u':
      STR2AVAL(auth, optarg);
      break;
    case 'C': {
      AVal av;
      STR2AVAL(av, optarg);
      if (!RTMP_SetOpt(&rtmp, &av_conn, &av))
        {
          RTMP_Log(RTMP_LOGERROR, "Invalid AMF parameter: %s", optarg);
          return RD_FAILED;
        }
      }
      break;
    case 'm':
      timeout = atoi(optarg);
      break;
    case 'A':
      dStartOffset = (int) (atof(optarg) * 1000.0);
      break;
    case 'B':
      dStopOffset = (int) (atof(optarg) * 1000.0);
      break;
    case 'T': {
      AVal token;
      STR2AVAL(token, optarg);
      RTMP_SetOpt(&rtmp, &av_token, &token);
      }
      break;
    case '#':
      bHashes = TRUE;
      break;
    case 'q':
      RTMP_debuglevel = RTMP_LOGCRIT;
      break;
    case 'V':
      RTMP_debuglevel = RTMP_LOGDEBUG;
      break;
    case 'z':
      RTMP_debuglevel = RTMP_LOGALL;
      break;
    case 'S':
      STR2AVAL(sockshost, optarg);
      break;
    default:
      RTMP_LogPrintf("unknown option: %c\n", opt);
      usage(argv[0]);
      return RD_FAILED;
      break;
    }
    }

  if (!hostname.av_len)
    {
      RTMP_Log(RTMP_LOGERROR,
      "您必须指定 主机名(hostname) (--host) 或 url (-r \"rtmp://host[:port]/playpath\") 包含 a hostname");
      return RD_FAILED;
    }
  if (playpath.av_len == 0)
    {
      RTMP_Log(RTMP_LOGERROR,
      "您必须指定 播放路径(playpath) (--playpath) 或 url (-r \"rtmp://host[:port]/playpath\") 包含 a playpath");
      return RD_FAILED;
    }

  if (protocol == RTMP_PROTOCOL_UNDEFINED)
    {
      RTMP_Log(RTMP_LOGWARNING,
      "您没有指定 协议(protocol) (--protocol) 或 rtmp url (-r), 默认协议 RTMP");
      protocol = RTMP_PROTOCOL_RTMP;
    }
  if (port == -1)
    {
      RTMP_Log(RTMP_LOGWARNING,
      "您没有指定 端口(port) (--port) 或 rtmp url (-r), 默认端口 1935");
      port = 0;
    }
  if (port == 0)
    {
      if (protocol & RTMP_FEATURE_SSL)
    port = 443;
      else if (protocol & RTMP_FEATURE_HTTP)
    port = 80;
      else
    port = 1935;
    }
```

```
    if (flvFile == 0)
      {
        RTMP_Log(RTMP_LOGWARNING,
        "请指定一个输出文件 (-o filename), using stdout");
        bStdoutMode = TRUE;
      }

    if (bStdoutMode && bResume)
      {
        RTMP_Log(RTMP_LOGWARNING,
        "Can't resume in stdout mode, ignoring --resume option");
        bResume = FALSE;
      }

    if (bLiveStream && bResume)
      {
        RTMP_Log(RTMP_LOGWARNING, "Can't resume live stream, ignoring --resume option");
        bResume = FALSE;
      }

  #ifdef CRYPTO
    if (swfVfy)
      {
        if (RTMP_HashSWF(swfUrl.av_val, (unsigned int *)&swfSize, hash, swfAge) == 0)
          {
            swfHash.av_val = (char *)hash;
            swfHash.av_len = RTMP_SWF_HASHLEN;
          }
      }

    if (swfHash.av_len == 0 && swfSize > 0)
      {
        RTMP_Log(RTMP_LOGWARNING,
        "Ignoring SWF size, supply also the hash with --swfhash");
        swfSize = 0;
      }

    if (swfHash.av_len != 0 && swfSize == 0)
      {
        RTMP_Log(RTMP_LOGWARNING,
        "Ignoring SWF hash, supply also the swf size  with --swfsize");
        swfHash.av_len = 0;
        swfHash.av_val = NULL;
      }
  #endif

    if (tcUrl.av_len == 0)
      {
        char str[512] = { 0 };

        tcUrl.av_len = snprintf(str, 511, "%s://%.*s:%d/%.*s",
             RTMPProtocolStringsLower[protocol], hostname.av_len,
             hostname.av_val, port, app.av_len, app.av_val);
        tcUrl.av_val = (char *) malloc(tcUrl.av_len + 1);
        strcpy(tcUrl.av_val, str);
      }

    int first = 1;

    // User defined seek offset
    if (dStartOffset > 0)
      {
        //直播流
        if (bLiveStream)
      {
        RTMP_Log(RTMP_LOGWARNING,
            "Can't seek in a live stream, ignoring --start option");
        dStartOffset = 0;
      }
      }
    //----------------
    rtmp.dlg->AppendCInfo("开始初始化RTMP连接的参数...");
    //----------------
    //设置
    RTMP_SetupStream(&rtmp, protocol, &hostname, port, &sockshost, &playpath,
             &tcUrl, &swfUrl, &pageUrl, &app, &auth, &swfHash, swfSize,
             &flashVer, &subscribepath, dSeek, dStopOffset, bLiveStream, timeout);
    //此处设置参数----------------
    rtmp.dlg->AppendCInfo("成功初始化RTMP连接的参数");
    //----------------------------
    char *temp=(char *)malloc(MAX_URL_LENGTH);

    memcpy(temp,rtmp.Link.hostname.av_val,rtmp.Link.hostname.av_len);
    temp[rtmp.Link.hostname.av_len]='\0';
    rtmp.dlg->AppendB_R_L_Info("主机名",temp);

    itoa(rtmp.Link.port,temp,10);
    rtmp.dlg->AppendB_R_L_Info("端口号",temp);

    memcpy(temp,rtmp.Link.app.av_val,rtmp.Link.app.av_len);
    temp[rtmp.Link.app.av_len]='\0';
```

```
  temp[rtmp.Link.app.av_len]='\0';
  rtmp.dlg->AppendB_R_L_Info("应用程序",temp);

  memcpy(temp,rtmp.Link.playpath.av_val,rtmp.Link.playpath.av_len);
  temp[rtmp.Link.playpath.av_len]='\0';
  rtmp.dlg->AppendB_R_L_Info("路径",temp);


  //----------------------------

  /* Try to keep the stream moving if it pauses on us */
  if (!bLiveStream && !(protocol & RTMP_FEATURE_HTTP))
    rtmp.Link.lFlags |= RTMP_LF_BUFX;

  off_t size = 0;

  // ok,我们必须获得timestamp of the last keyframe (only keyframes are seekable) / last audio frame (audio only streams)
  if (bResume)
    {
    //打开文件，输出的文件(Resume)
      nStatus =
    OpenResumeFile(flvFile, &file, &size, &metaHeader, &nMetaHeaderSize,
               &duration);
      if (nStatus == RD_FAILED)
    goto clean;

      if (!file)
    {
      // file does not exist, so go back into normal mode
      bResume = FALSE;  // we are back in fresh file mode (otherwise finalizing file won't be done)
    }
      else
    {
    //获取最后一个关键帧
      nStatus = GetLastKeyframe(file, nSkipKeyFrames,
                     &dSeek, &initialFrame,
                     &initialFrameType, &nInitialFrameSize);
      if (nStatus == RD_FAILED)
        {
          RTMP_Log(RTMP_LOGDEBUG, "Failed to get last keyframe.");
          goto clean;
        }

      if (dSeek == 0)
        {
          RTMP_Log(RTMP_LOGDEBUG,
          "Last keyframe is first frame in stream, switching from resume to normal mode!");
          bResume = FALSE;
        }
    }
    }
  //如果输出文件不存在
  if (!file)
    {
      if (bStdoutMode)
    {
    //直接输出到stdout
      file = stdout;
      SET_BINMODE(file);
    }
      else
    {
    //打开一个文件
    //w+b 读写打开或建立一个二进制文件，允许读和写。
        //-----------------
        rtmp.dlg->AppendCInfo("创建输出文件...");
        //---------------------------
      file = fopen(flvFile, "w+b");
      if (file == 0)
        {
            //-----------------
            rtmp.dlg->AppendCInfo("创建输出文件失败！");
            //---------------------------
          RTMP_LogPrintf("Failed to open file! %s\n", flvFile);
          return RD_FAILED;
        }
      rtmp.dlg->AppendCInfo("成功创建输出文件");
    }
    }

#ifdef _DEBUG
  netstackdump = fopen("netstackdump", "wb");
  netstackdump_read = fopen("netstackdump_read", "wb");
#endif

  while (!RTMP_ctrlC)
    {
      RTMP_Log(RTMP_LOGDEBUG, "Setting buffer time to: %dms", bufferTime);
      //设置Buffer时间
      //-----------------
      rtmp.dlg->AppendCInfo("设置缓冲(Buffer)的时间");
      //---------------------------
```

```
    RTMP_SetBufferMS(&rtmp, bufferTime);
    //第一次执行
    if (first)
  {
    first = 0;
    RTMP_LogPrintf("开始建立连接！\n");
    //-----------------
    rtmp.dlg->AppendCInfo("开始建立连接（NetConnection）...");
    //----------------------------
    //建立连接(Connect)
    if (!RTMP_Connect(&rtmp, NULL))
      {
          //-----------------
          rtmp.dlg->AppendCInfo("建立连接（NetConnection）失败！");
          //----------------------------
        nStatus = RD_FAILED;
        break;
      }
    //-----------------
    rtmp.dlg->AppendCInfo("成功建立连接（NetConnection）");
    //----------------------------
    //RTMP_Log(RTMP_LOGINFO, "已链接...");

    // User defined seek offset
    if (dStartOffset > 0)
      {
        // Don't need the start offset if resuming an existing file
        if (bResume)
      {
        RTMP_Log(RTMP_LOGWARNING,
            "Can't seek a resumed stream, ignoring --start option");
        dStartOffset = 0;
      }
        else
      {
        dSeek = dStartOffset;
      }
      }

    // Calculate the length of the stream to still play
    if (dStopOffset > 0)
      {
        // Quit if start seek is past required stop offset
        if (dStopOffset <= dSeek)
      {
        RTMP_LogPrintf("Already Completed\n");
        nStatus = RD_SUCCESS;
        break;
      }
      }
    //创建流(Stream)（发送connect命令消息后处理传来的数据）
    itoa(rtmp.m_inChunkSize,temp,10);
    rtmp.dlg->AppendB_R_Info("输入Chunk大小",temp);
    itoa(rtmp.m_outChunkSize,temp,10);
    rtmp.dlg->AppendB_R_Info("输出Chunk大小",temp);
    itoa(rtmp.m_stream_id,temp,10);
    rtmp.dlg->AppendB_R_Info("Stream ID",temp);
    itoa(rtmp.m_nBufferMS,temp,10);
    rtmp.dlg->AppendB_R_Info("Buffer时长（ms）",temp);
    itoa(rtmp.m_nServerBW,temp,10);
    rtmp.dlg->AppendB_R_Info("ServerBW",temp);
    itoa(rtmp.m_nClientBW,temp,10);
    rtmp.dlg->AppendB_R_Info("ClientBW",temp);
    itoa((int)rtmp.m_fEncoding,temp,10);
    rtmp.dlg->AppendB_R_Info("命令消息编码方法",temp);
    itoa((int)rtmp.m_fDuration,temp,10);
    rtmp.dlg->AppendB_R_Info("时长（s）",temp);

    rtmp.dlg->ShowBInfo();
    free(temp);
    //-----------------
    rtmp.dlg->AppendCInfo("开始建立网络流（NetStream）");
    //----------------------------
    if (!RTMP_ConnectStream(&rtmp, dSeek))
      {
      //-----------------
      rtmp.dlg->AppendCInfo("建立网络流（NetStream）失败！");
      //-----------------
        nStatus = RD_FAILED;
        break;
      }
    //-----------------
    rtmp.dlg->AppendCInfo("成功建立网络流（NetStream）！");
    //-----------------
  }
    else
  {
    nInitialFrameSize = 0;

        if (retries)
          {
```

```
          RTMP_Log(RTMP_LOGERROR, "Failed to resume the stream\n\n");
          if (!RTMP_IsTimedout(&rtmp))
            nStatus = RD_FAILED;
          else
            nStatus = RD_INCOMPLETE;
          break;
            }
      RTMP_Log(RTMP_LOGINFO, "Connection timed out, trying to resume.\n\n");
          /* Did we already try pausing, and it still didn't work? */
          if (rtmp.m_pausing == 3)
            {
              /* Only one try at reconnecting... */
              retries = 1;
              dSeek = rtmp.m_pauseStamp;
              if (dStopOffset > 0)
                {
                  if (dStopOffset <= dSeek)
                    {
                      RTMP_LogPrintf("Already Completed\n");
              nStatus = RD_SUCCESS;
              break;
                    }
                }
              if (!RTMP_ReconnectStream(&rtmp, dSeek))
                {
              RTMP_Log(RTMP_LOGERROR, "Failed to resume the stream\n\n");
              if (!RTMP_IsTimedout(&rtmp))
            nStatus = RD_FAILED;
              else
            nStatus = RD_INCOMPLETE;
              break;
                }
            }
      else if (!RTMP_ToggleStream(&rtmp))
        {
          RTMP_Log(RTMP_LOGERROR, "Failed to resume the stream\n\n");
          if (!RTMP_IsTimedout(&rtmp))
        nStatus = RD_FAILED;
          else
        nStatus = RD_INCOMPLETE;
          break;
        }
      bResume = TRUE;
    }
    //----------------

    //----------------
    rtmp.dlg->AppendCInfo("开始将媒体数据写入文件");
    //----------------
      //下载,写入文件
      nStatus = Download(&rtmp, file, dSeek, dStopOffset, duration, bResume,
             metaHeader, nMetaHeaderSize, initialFrame,
             initialFrameType, nInitialFrameSize,
             nSkipKeyFrames, bStdoutMode, bLiveStream, bHashes,
             bOverrideBufferTime, bufferTime, &percent);
      free(initialFrame);
      initialFrame = NULL;

      /* If we succeeded, we're done.
       */
      if (nStatus != RD_INCOMPLETE || !RTMP_IsTimedout(&rtmp) || bLiveStream)
    break;
    }
    //当下载完的时候
  if (nStatus == RD_SUCCESS)
    {
        //----------------
        rtmp.dlg->AppendCInfo("写入文件完成");
        //----------------
      RTMP_LogPrintf("Download complete\n");
    }
    //没下载完的时候
  else if (nStatus == RD_INCOMPLETE)
    {
        //----------------
        rtmp.dlg->AppendCInfo("写入文件可能不完整");
        //----------------
      RTMP_LogPrintf
    ("Download may be incomplete (downloaded about %.2f%%), try resuming\n",
     percent);
    }
  //后续清理工作
clean:
  //----------------
  rtmp.dlg->AppendCInfo("关闭连接");
  //----------------
  RTMP_Log(RTMP_LOGDEBUG, "Closing connection.\n");
  RTMP_Close(&rtmp);
  rtmp.dlg->AppendCInfo("关闭文件");
  if (file != 0)
    fclose(file);
```

```
  rtmp.dlg->AppendCInfo("关闭Socket");
  CleanupSockets();

#ifdef _DEBUG
  if (netstackdump != 0)
    fclose(netstackdump);
  if (netstackdump_read != 0)
    fclose(netstackdump_read);
#endif
  return nStatus;
}
```

其中InitSocket()代码很简单，初始化了Socket，如下：

[cpp]

```
// 初始化 sockets
int
InitSockets()
{
#ifdef WIN32
  WORD version;
  WSADATA wsaData;

  version = MAKEWORD(1, 1);
  return (WSAStartup(version, &wsaData) == 0);
#else
  return TRUE;
#endif
}
```

CleanupSockets()则更简单：

[cpp]

```
inline void
CleanupSockets()
{
#ifdef WIN32
  WSACleanup();
#endif
}
```

Download()函数则比较复杂：

[cpp]

```
int
Download(RTMP * rtmp,       // connected RTMP object
     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]
{
  int32_t now, lastUpdate;
  int bufferSize = 64 * 1024;
  char *buffer = (char *) malloc(bufferSize);
  int nRead = 0;

  //long ftell(FILE *stream);
  //返回当前文件指针
  RTMP_LogPrintf("开始下载！\n");
  off_t size = ftello(file);
  unsigned long lastPercent = 0;
  //时间戳
  rtmp->m_read.timestamp = dSeek;

  *percent = 0.0;

  if (rtmp->m_read.timestamp)
    {
      RTMP_Log(RTMP_LOGDEBUG, "Continuing at TS: %d ms\n", rtmp->m_read.timestamp);
    }
  //是直播
  if (bLiveStream)
    {
      RTMP_LogPrintf("直播流\n");
    }
  else
    {
      // print initial status
      // Workaround to exit with 0 if the file is fully (> 99.9%) downloaded
      if (duration > 0)
    {
      if ((double) rtmp->m_read.timestamp >= (double) duration * 999.0)
        {
          RTMP_LogPrintf("Already Completed at: %.3f sec Duration=%.3f sec\n",
            (double) rtmp->m_read.timestamp / 1000.0,
            (double) duration / 1000.0);
```

```
            return RD_SUCCESS;
          }
        else
          {
            *percent = ((double) rtmp->m_read.timestamp) / (duration * 1000.0) * 100.0;
            *percent = ((double) (int) (*percent * 10.0)) / 10.0;
            RTMP_LogPrintf("%s download at: %.3f kB / %.3f sec (%.1f%%)\n",
              bResume ? "Resuming" : "Starting",
              (double) size / 1024.0, (double) rtmp->m_read.timestamp / 1000.0,
              *percent);
          }
      }
        else
      {
        RTMP_LogPrintf("%s download at: %.3f kB\n",
              bResume ? "Resuming" : "Starting",
              (double) size / 1024.0);
      }
      }

    if (dStopOffset > 0)
      RTMP_LogPrintf("For duration: %.3f sec\n", (double) (dStopOffset - dSeek) / 1000.0);

    //各种设置参数到rtmp连接
    if (bResume && nInitialFrameSize > 0)
    rtmp->m_read.flags |= RTMP_READ_RESUME;
    rtmp->m_read.initialFrameType = initialFrameType;
    rtmp->m_read.nResumeTS = dSeek;
    rtmp->m_read.metaHeader = metaHeader;
    rtmp->m_read.initialFrame = initialFrame;
    rtmp->m_read.nMetaHeaderSize = nMetaHeaderSize;
    rtmp->m_read.nInitialFrameSize = nInitialFrameSize;

    now = RTMP_GetTime();
    lastUpdate = now - 1000;
    do
      {
      //从rtmp中把bufferSize（64k）个数据读入buffer
        nRead = RTMP_Read(rtmp, buffer, bufferSize);
        //RTMP_LogPrintf("nRead: %d\n", nRead);
        if (nRead > 0)
      {
      //函数：size_t fwrite(const void* buffer,size_t size,size_t count,FILE* stream);
      //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
      //（1）buffer：是一个指针，对fwrite来说，是要输出数据的地址。
      //（2）size：要写入内容的单字节数；
      //（3）count:要进行写入size字节的数据项的个数；
      //（4）stream:目标文件指针。
      //（5）返回实际写入的数据项个数count。
      //关键。把buffer里面的数据写成文件
          if (fwrite(buffer, sizeof(unsigned char), nRead, file) !=
            (size_t) nRead)
          {
            RTMP_Log(RTMP_LOGERROR, "%s: Failed writing, exiting!", __FUNCTION__);
            free(buffer);
            return RD_FAILED;
          }
          //记录已经写入的字节数
        size += nRead;

        //RTMP_LogPrintf("write %dbytes (%.1f kB)\n", nRead, nRead/1024.0);
        if (duration <= 0) // if duration unknown try to get it from the stream (onMetaData)
          duration = RTMP_GetDuration(rtmp);

        if (duration > 0)
          {
            // make sure we claim to have enough buffer time!
            if (!bOverrideBufferTime && bufferTime < (duration * 1000.0))
          {
            bufferTime = (uint32_t) (duration * 1000.0) + 5000;   // 再加5s以确保buffertime足够长

            RTMP_Log(RTMP_LOGDEBUG,
                "Detected that buffer time is less than duration, resetting to: %dms",
                bufferTime);
            //重设Buffer长度
            RTMP_SetBufferMS(rtmp, bufferTime);
            //给服务器发送UserControl消息通知Buffer改变
            RTMP_UpdateBufferMS(rtmp);
          }
            //计算百分比
            *percent = ((double) rtmp->m_read.timestamp) / (duration * 1000.0) * 100.0;
            *percent = ((double) (int) (*percent * 10.0)) / 10.0;
            if (bHashes)
          {
            if (lastPercent + 1 <= *percent)
              {
                RTMP_LogStatus("#");
                lastPercent = (unsigned long) *percent;
              }
          }
            else
```

```
        {
            //设置显示数据的更新间隔200ms
          now = RTMP_GetTime();
          if (abs(now - lastUpdate) > 200)
            {
              RTMP_LogStatus("\r%.3f kB / %.2f sec (%.1f%%)",
                (double) size / 1024.0,
                (double) (rtmp->m_read.timestamp) / 1000.0, *percent);
              lastUpdate = now;
            }
        }
        }
      else
        {
        //现在距离开机的毫秒数
          now = RTMP_GetTime();
          //每间隔200ms刷新一次数据
          if (abs(now - lastUpdate) > 200)
        {
          if (bHashes)
            RTMP_LogStatus("#");
          else
            //size为已写入文件的字节数
            RTMP_LogStatus("\r%.3f kB / %.2f sec", (double) size / 1024.0,
                  (double) (rtmp->m_read.timestamp) / 1000.0);
          lastUpdate = now;
        }
        }
    }
#ifdef _DEBUG
      else
    {
      RTMP_Log(RTMP_LOGDEBUG, "zero read!");
    }
#endif

    }
  while (!RTMP_ctrlC && nRead > -1 && RTMP_IsConnected(rtmp) && !RTMP_IsTimedout(rtmp));
  free(buffer);
  if (nRead < 0)
    //nRead是读取情况
    nRead = rtmp->m_read.status;

  /* Final status update */
  if (!bHashes)
    {
      if (duration > 0)
    {
      *percent = ((double) rtmp->m_read.timestamp) / (duration * 1000.0) * 100.0;
      *percent = ((double) (int) (*percent * 10.0)) / 10.0;
      //输出
      RTMP_LogStatus("\r%.3f kB / %.2f sec (%.1f%%)",
        (double) size / 1024.0,
        (double) (rtmp->m_read.timestamp) / 1000.0, *percent);
    }
      else
    {
      RTMP_LogStatus("\r%.3f kB / %.2f sec", (double) size / 1024.0,
        (double) (rtmp->m_read.timestamp) / 1000.0);
    }
    }

  RTMP_Log(RTMP_LOGDEBUG, "RTMP_Read returned: %d", nRead);
  //读取错误
  if (bResume && nRead == -2)
    {
      RTMP_LogPrintf("Couldn't resume FLV file, try --skip %d\n\n",
        nSkipKeyFrames + 1);
      return RD_FAILED;
    }
  //读取正确
  if (nRead == -3)
    return RD_SUCCESS;
  //没读完...
  if ((duration > 0 && *percent < 99.9) || RTMP_ctrlC || nRead < 0
      || RTMP_IsTimedout(rtmp))
    {
      return RD_INCOMPLETE;
    }

  return RD_SUCCESS;
}
```

以上内容是我能理解到的rtmpdump.c里面的内容。

rtmpdump源代码（Linux）：http://download.csdn.net/detail/leixiaohua1020/6376561 rtmpdump源代码（VC 2005 工程）：http://download.csdn.net/detail/leixiaohua1020/6563163

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