

RTSP服务运作

1. RTSP首先需建立TCP侦听socket。可见于此函数：

```
1 rtspServer = DynamicRTSPServer::createNew(*env, rtspServerPortNum, authDB);
2
3 DynamicRTSPServer*
4 DynamicRTSPServer::createNew(UsageEnvironment& env, Port ourPort,
5     UserAuthenticationDatabase* authDatabase,
6     unsigned reclamationTestSeconds) {
7     int ourSocket = setUpOurSocket(env, ourPort);
8     if (ourSocket == -1) return NULL;
9
10    return new DynamicRTSPServer(env, ourSocket, ourPort, authDatabase, reclamationTestSeconds);
11 }
```

- setUpOurSocket

```
1 setUpOurSocket()
2     ->setupStreamSocket() 建立套接字
3     ->listen()
4     ->getSourcePort() //bind()
```

2. 运行：rtspServer = DynamicRTSPServer::createNew(*env, rtspServerPortNum, authDB); 会运行DynamicRTSPServer的构造函数

```
1 DynamicRTSPServer::DynamicRTSPServer(UsageEnvironment& env, int ourSocket,
2     Port ourPort,
3     UserAuthenticationDatabase* authDatabase, unsigned reclamationTestSeconds)
4     : RTSPServer(env, ourSocket, ourPort, authDatabase, reclamationTestSeconds) {
5 }
```

• RTSPServer(): 要侦听客户端的连接，就需要利用任务调度机制了，所以需添加一个socket handler。可见于此函数：

```
1 RTSPServer::RTSPServer(UsageEnvironment& env,
2     int ourSocket, Port ourPort,
3     UserAuthenticationDatabase* authDatabase,
4     unsigned reclamationSeconds)
5     : GenericMediaServer(env, ourSocket, ourPort, reclamationSeconds),
6     fHTTPServerSocket(-1), fHTTPServerPort(0),
```

```

7     fClientConnectionsForHTTPTunneling(NULL), // will get created if needed
8     fTCPStreamingDatabase(HashTable::create(ONE_WORD_HASH_KEYS)),
9     fPendingRegisterOrDeregisterRequests(HashTable::create(ONE_WORD_HASH_KEYS)),
10    fRegisterOrDeregisterRequestCounter(0), fAuthDB(authDatabase), fAllowStreamingRTPOver
11 }

```

- GenericMediaServer:通过这个函数来添加socket handler

```

1  GenericMediaServer
2  ::GenericMediaServer(UsageEnvironment& env, int ourSocket, Port ourPort,
3      unsigned reclamationSeconds)
4  : Medium(env),
5    fServerSocket(ourSocket), fServerPort(ourPort), fReclamationSeconds(reclamationSeconds),
6    fServerMediaSessions(HashTable::create(StringHashKeys)),
7    fClientConnections(HashTable::create(OneWordHashKeys)),
8    fClientSessions(HashTable::create(StringHashKeys)),
9    fPreviousClientId(0)
10 {
11     ignoreSigPipeOnSocket(fServerSocket); // so that clients on the same host that are killed
12
13     // Arrange to handle connections from others:
14     env.taskScheduler().turnOnBackgroundReadHandling(fServerSocket, incomingConnectionHandler);
15 }

```

- env.taskScheduler().turnOnBackgroundReadHandling(fServerSocket, incomingConnectionHandler, this);

```

1  这个函数其实还是调用的
2  void turnOnBackgroundReadHandling(int socketNum, BackgroundHandlerProc* handlerProc, void* clientData);
3      setBackgroundHandling(socketNum, SOCKET_READABLE, handlerProc, clientData);
4  }
5  handlerProc : 是select检测到函数动作之后的处理函数。

```

- incomingConnectionHandler

```

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- setBackgroundHandling () : 向任务调度对象添加socket Task任务

3. 当收到客户的连接时需保存下代表客户端的新socket, 以后用这个socket与这个客户通讯。每个客户将来会对应一个rtp会话, 而且各客户的RTSP请求只控制自己的rtp会话, 那么最好建立一个会话

类，代表各客户的rtsp会话。于是类RTSPServer::RTSPClientSession产生，它保存的代表客户的socket。下为RTSPClientSession的创建过程

这个在新版本怎么调用的还没理清

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4. Client会话类还没有理清清楚

5. ServerMediaSession::createNew是一个简单工厂模式函数，在其中new了一个ServerMediaSession对象，看一下ServerMediaSession这个类的定义。

```
1
2 class ServerMediaSession: public Medium {
3 public:
4     static ServerMediaSession* createNew(UsageEnvironment& env,
5         char const* streamName = NULL,
6         char const* info = NULL,
7         char const* description = NULL,
8         Boolean isSSM = False,
9         char const* miscSDPLines = NULL);
10
11     static Boolean lookupByName(UsageEnvironment& env,
12         char const* mediumName,
13         ServerMediaSession*& resultSession);
14
15     char* generateSDPDescription(int addressFamily); // based on the entire session /产生媒
16         // Note: The caller is responsible for freeing the returned string
17
18     char const* streamName() const { return fStreamName; } //返回流的名称
19
20     Boolean addSubsession(ServerMediaSubsession* subsession); //添加表示子会话的Server
21     unsigned numSubsessions() const { return fSubsessionCounter; }
22
23     void testScaleFactor(float& scale); // sets "scale" to the actual supported scale //返
24     float duration() const;
25         // a result == 0 means an unbounded session (the default)
26         // a result < 0 means: subsession durations differ; the result is -(the largest).
27         // a result > 0 means: this is the duration of a bounded session
28
29     virtual void noteLiveness();
30
31     // called whenever a client - accessing this media - notes liveness.
```

```

31     // The default implementation does nothing, but subclasses can redefine this - e.g.,
32     // want to remove long-unused "ServerMediaSession"s from the server.
33
34     unsigned referenceCount() const { return fReferenceCount; } //返回请求该流的RTSP客户端数
35     void incrementReferenceCount() { ++fReferenceCount; }
36     void decrementReferenceCount() { if (fReferenceCount > 0) --fReferenceCount; }
37     Boolean& deleteWhenUnreferenced() { return fDeleteWhenUnreferenced; } //fDeleteWhenUnre-
38
39     void deleteAllSubsessions();
40     // Removes and deletes all subsessions added by "addSubsession()", returning us to an
41     // Note: If you have already added this "ServerMediaSession" to a "RTSPServer" then,
42     // you must first close any client connections that use it,
43     // by calling "RTSPServer::closeAllClientSessionsForServerMediaSession()".
44
45 protected:
46     ServerMediaSession(UsageEnvironment& env, char const* streamName,
47         char const* info, char const* description,
48         Boolean isSSM, char const* miscSDPLines);
49     // called only by "createNew()"
50
51     virtual ~ServerMediaSession();
52
53 private: // redefined virtual functions
54     virtual Boolean isServerMediaSession() const;
55
56 private:
57     Boolean fIsSSM;
58
59     // Linkage fields:
60     friend class ServerMediaSubsessionIterator; // ServerMediaSubsessionIterator是一个用于访
61     ServerMediaSubsession* fSubsessionsHead;
62     ServerMediaSubsession* fSubsessionsTail;
63     unsigned fSubsessionCounter;
64
65     char* fStreamName;
66     char* fInfoSDPString;
67     char* fDescriptionSDPString;
68     char* fMiscSDPLines;
69
70     struct timeval fCreationTime;

```

```

70 unsigned fReferenceCount;
71 Boolean fDeleteWhenUnreferenced;
72 };

```

6. 看一下ServerMediaSubsession这个类。ServerMediaSession原先说代表一个流，其实是不准确的。它代表的是server端的一个媒体的名字，而说ServerMediaSubsession代表一个Track是准确的。以后流指的是那些有数据流动的组合。

```

1
2 class ServerMediaSubsession: public Medium {
3 public:
4     unsigned trackNumber() const { return fTrackNumber; } // 每个ServerMediaSubsession又叫一个
5     char const* trackId(); // 产生关于该视频流或者音频流的描述
6     virtual char const* sdpLines() = 0;
7     virtual void getStreamParameters(unsigned sessionId, // in
8         struct sockaddr_storage const& clientAddress, // in
9         Port const& clientRTPPort, // in
10        Port const& clientRTCPPort, // in
11        int tcpSocketNum, // in (-1 means use UDP, not TCP)
12        unsigned char rtpChannelId, // in (used if TCP)
13        unsigned char rtcpChannelId, // in (used if TCP)
14        struct sockaddr_storage& destinationAddress, // in out
15        u_int8_t& destinationTTL, // in out
16        Boolean& isMulticast, // out
17        Port& serverRTPPort, // out
18        Port& serverRTCPPort, // out
19        void*& streamToken // out
20    ) = 0;
21     virtual void startStream(unsigned sessionId, void* streamToken, // 开始流化
22         TaskFunc* rtcpRRHandler,
23         void* rtcpRRHandlerClientData,
24         unsigned short& rtpSeqNum,
25         unsigned& rtpTimestamp,
26         ServerRequestAlternativeByteHandler* serverRequestAlternativeByteHandler,
27         void* serverRequestAlternativeByteHandlerClientData) = 0;
28     virtual void pauseStream(unsigned sessionId, void* streamToken); // 暂停流化
29     virtual void seekStream(unsigned sessionId, void* streamToken, double& seekNPT, //
30         double streamDuration, u_int64_t& numBytes);
31         // This routine is used to seek by relative (i.e., NPT) time.
32         // "streamDuration", if >0.0, specifies how much data to stream, past "seekNPT". (

```

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33     // "numBytes" returns the size (in bytes) of the data to be streamed, or 0 if unknown
34 virtual void seekStream(unsigned sessionId, void* streamToken, char*& absStart,
35     // This routine is used to seek by 'absolute' time.
36     // "absStart" should be a string of the form "YYYYMMDDTHHMMSSZ" or "YYYYMMDDTHHMMSS"
37     // "absEnd" should be either NULL (for no end time), or a string of the same form as "absStart"
38     // These strings may be modified in-place, or can be reassigned to a newly-allocated buffer
39 virtual void nullSeekStream(unsigned sessionId, void* streamToken,      //// Frame
40     double streamEndTime, u_int64_t& numBytes);
41     // Called whenever we're handling a "PLAY" command without a specified start time.
42 virtual void setStreamScale(unsigned sessionId, void* streamToken, float scale);
43 virtual float getCurrentNPT(void* streamToken);
44 virtual FramedSource* getStreamSource(void* streamToken);
45 virtual void getRTPSinkandRTCP(void* streamToken,
46     RTPSink const*& rtpSink, RTCPInstance const*& rtcp) = 0;
47     // Returns pointers to the "RTPSink" and "RTCPInstance" objects for "streamToken".
48     // (This can be useful if you want to get the associated 'Groupsock' objects, for example)
49     // You must not delete these objects, or start/stop playing them; instead, that is done by
50     // using the "startStream()" and "deleteStream()" functions.
51 virtual void deleteStream(unsigned sessionId, void*& streamToken);
52
53 virtual void testScaleFactor(float& scale); // sets "scale" to the actual supported scale factor
54 virtual float duration() const;
55     // returns 0 for an unbounded session (the default)
56     // returns > 0 for a bounded session
57 virtual void getAbsoluteTimeRange(char*& absStartTime, char*& absEndTime) const; //// :
58     // Subclasses can reimplement this iff they support seeking by 'absolute' time.
59
60 // The following may be called by (e.g.) SIP servers, for which the
61 // address and port number fields in SDP descriptions need to be non-zero:
62 void setServerAddressAndPortForSDP(struct sockaddr_storage const& address,
63     portNumBits portBits);
64
65 protected: // we're a virtual base class
66     ServerMediaSubsession(UsageEnvironment& env);
67     virtual ~ServerMediaSubsession();
68
69 char const* rangeSDPLine() const;
70     // returns a string to be delete[]d

```

```
72  ServerMediaSession* fParentSession;
73  struct sockaddr_storage fServerAddressForSDP;
74  portNumBits fPortNumForSDP;
75
76  private:
77      friend class ServerMediaSession;
78      friend class ServerMediaSubsessionIterator;
79      ServerMediaSubsession* fNext;
80
81      unsigned fTrackNumber; // within an enclosing ServerMediaSession
82      char const* fTrackId;
83  };
```

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