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Mela et al.

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(54) **STREAMING MEDIA**

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(51) **Int. Cl.**
H04L 12/66 (2006.01)

(52) **U.S. Cl.** **370/261**; 370/352; 709/203

(58) **Field of Classification Search** 370/260, 370/261, 341, 352, 423, 468, 395.2, 401, 370/465, 466; 395/200.61; 455/518, 519; 709/231, 203–204
See application file for complete search history.

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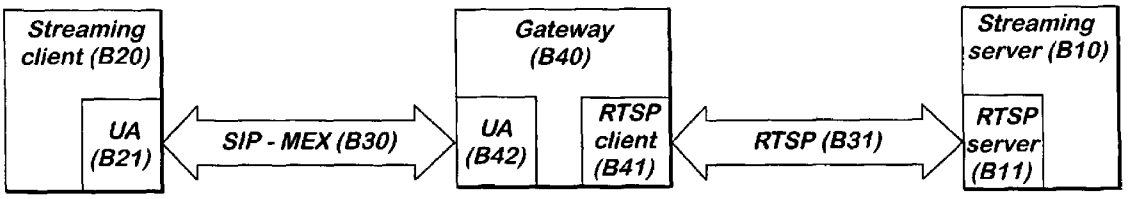
Assistant Examiner—Pao Sinkantarakorn

(57) **ABSTRACT**

A method for providing a session invitation protocol (SIP) session between a first and a second entity. An SIP session is established between the first and second entity. At least one media stream is transmitted from the first entity to the second entity. At least one of transmission, storage and play back of the at least one media stream, is controlled in the SIP session at the first and/or second entity.

27 Claims, 7 Drawing Sheets



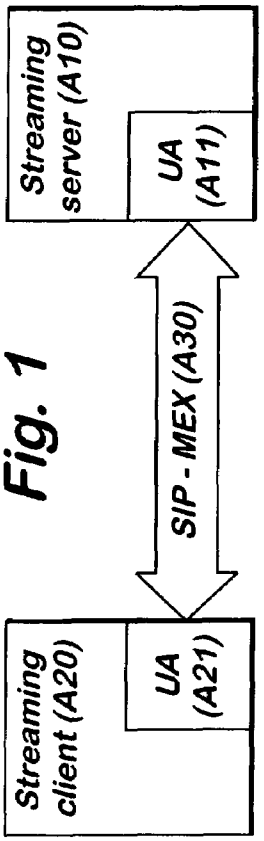


Fig. 1

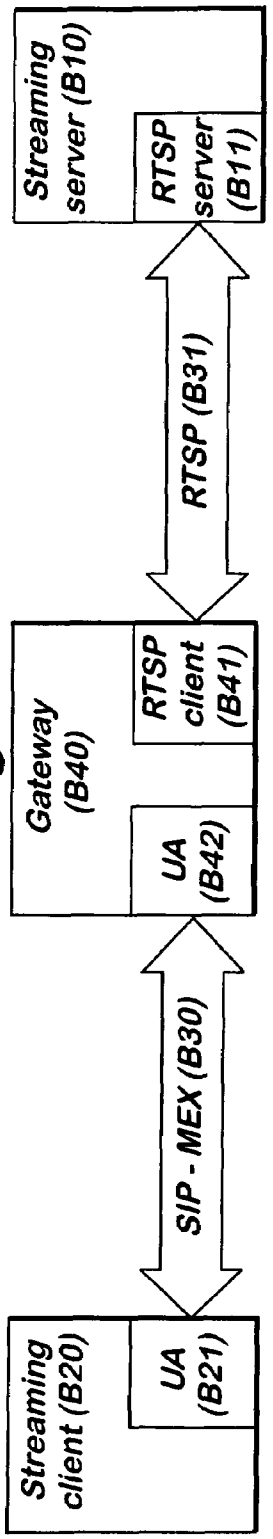


Fig. 2

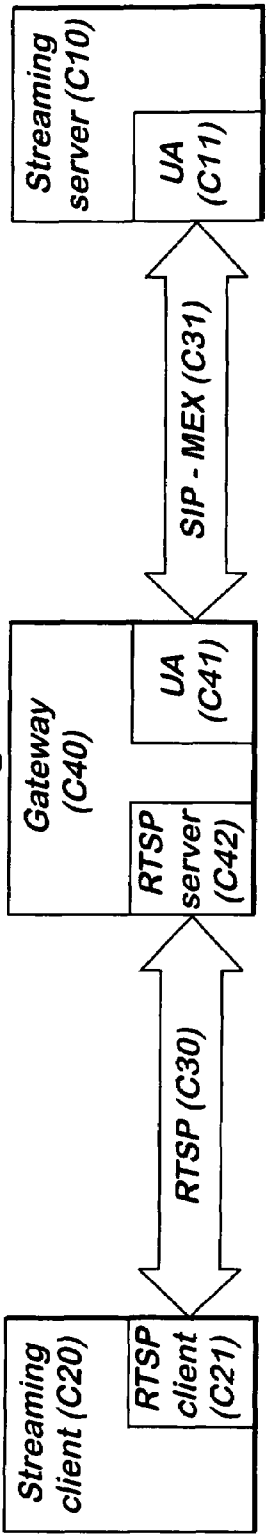


Fig. 3

Fig. 4

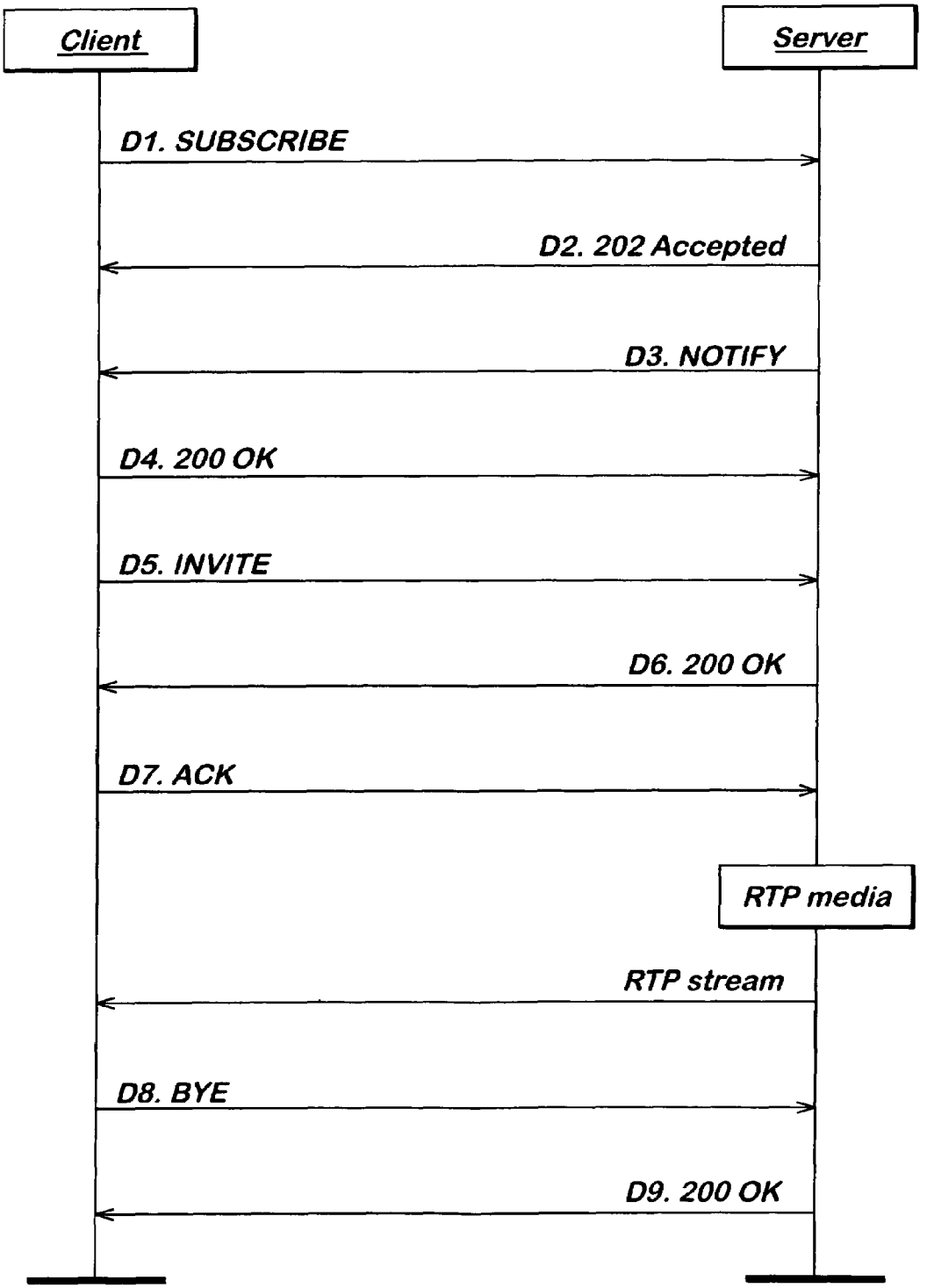

Client
Server
D1. SUBSCRIBE
D2. 202 Accepted
D3. NOTIFY
D4. 200 OK
D5. INVITE
D6. 200 OK
D7. ACK
RTP media
RTP stream
D8. BYE
D9. 200 OK

Fig. 5

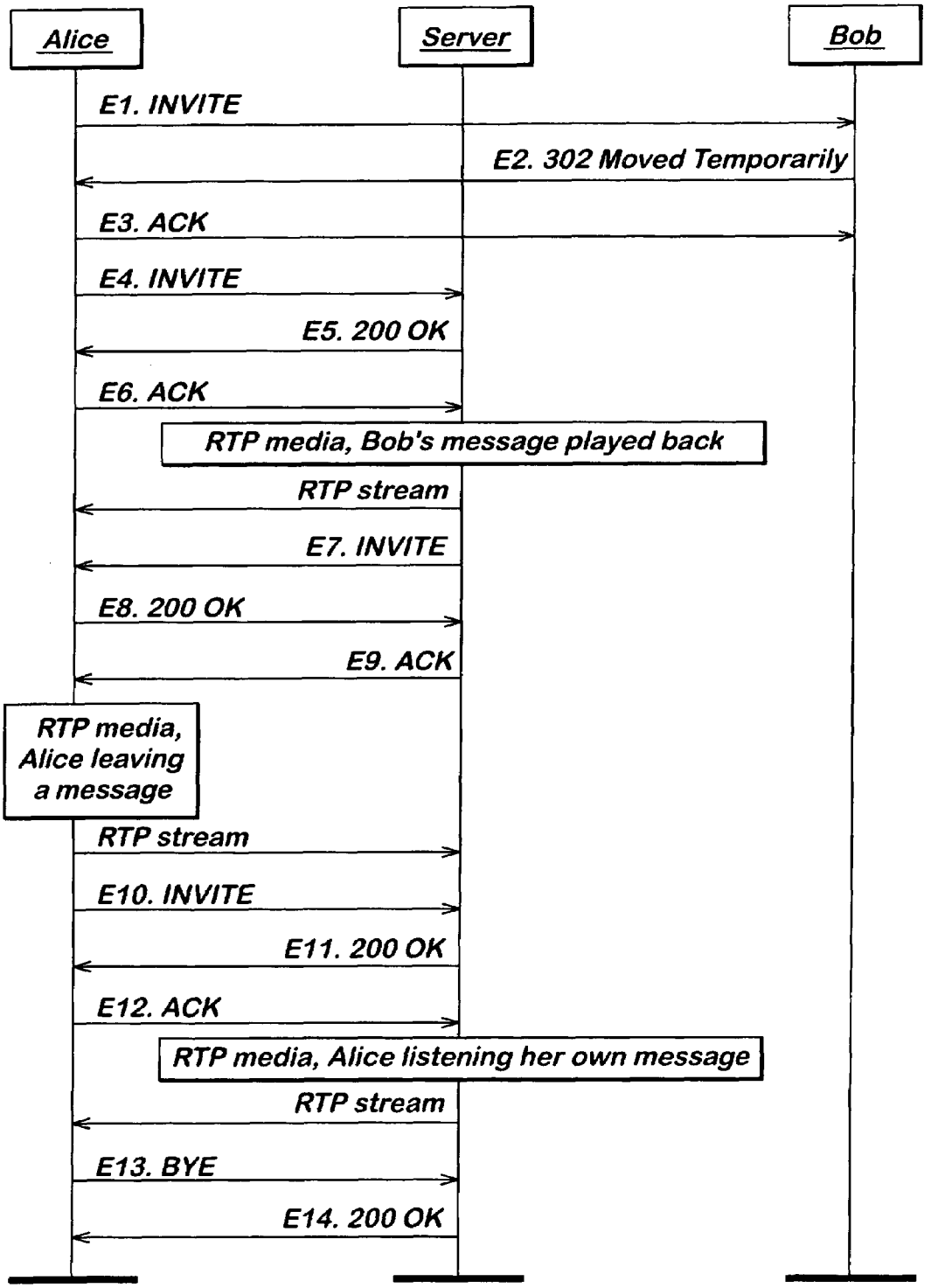

Alice
Server
Bob
E1. INVITE
E2. 302 Moved Temporarily
E3. ACK
E4. INVITE
E5. 200 OK
E6. ACK
RTP media, Bob's message played back
RTP stream
E7. INVITE
E8. 200 OK
E9. ACK
RTP media, Alice leaving a message
RTP stream
E10. INVITE
E11. 200 OK
E12. ACK
RTP media, Alice listening her own message
RTP stream
E13. BYE
E14. 200 OK

Fig. 6

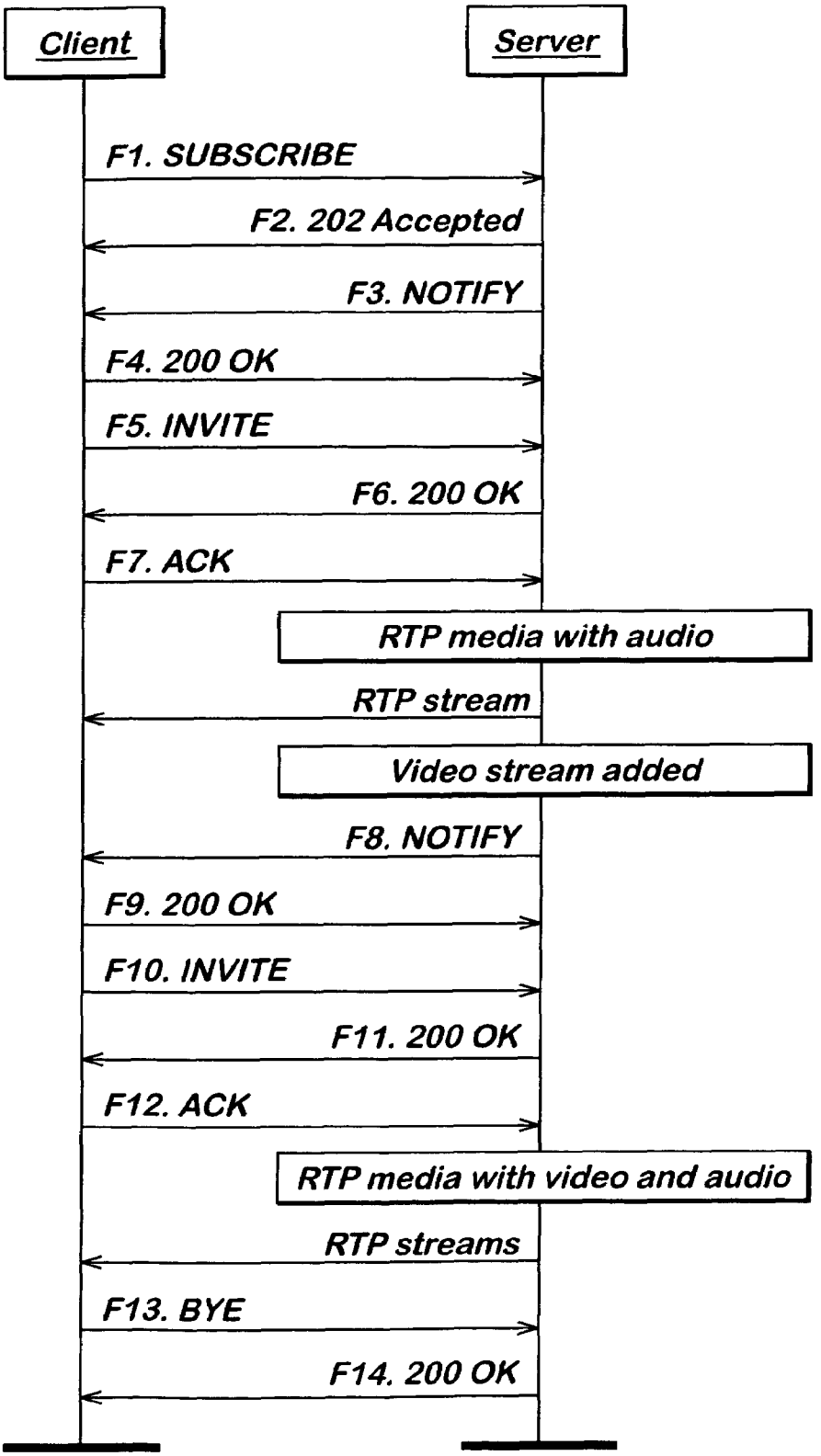

Client
Server
F1. SUBSCRIBE
F2. 202 Accepted
F3. NOTIFY
F4. 200 OK
F5. INVITE
F6. 200 OK
F7. ACK
RTP media with audio
RTP stream
Video stream added
F8. NOTIFY
F9. 200 OK
F10. INVITE
F11. 200 OK
F12. ACK
RTP media with video and audio
RTP streams
F13. BYE
F14. 200 OK

Fig. 7

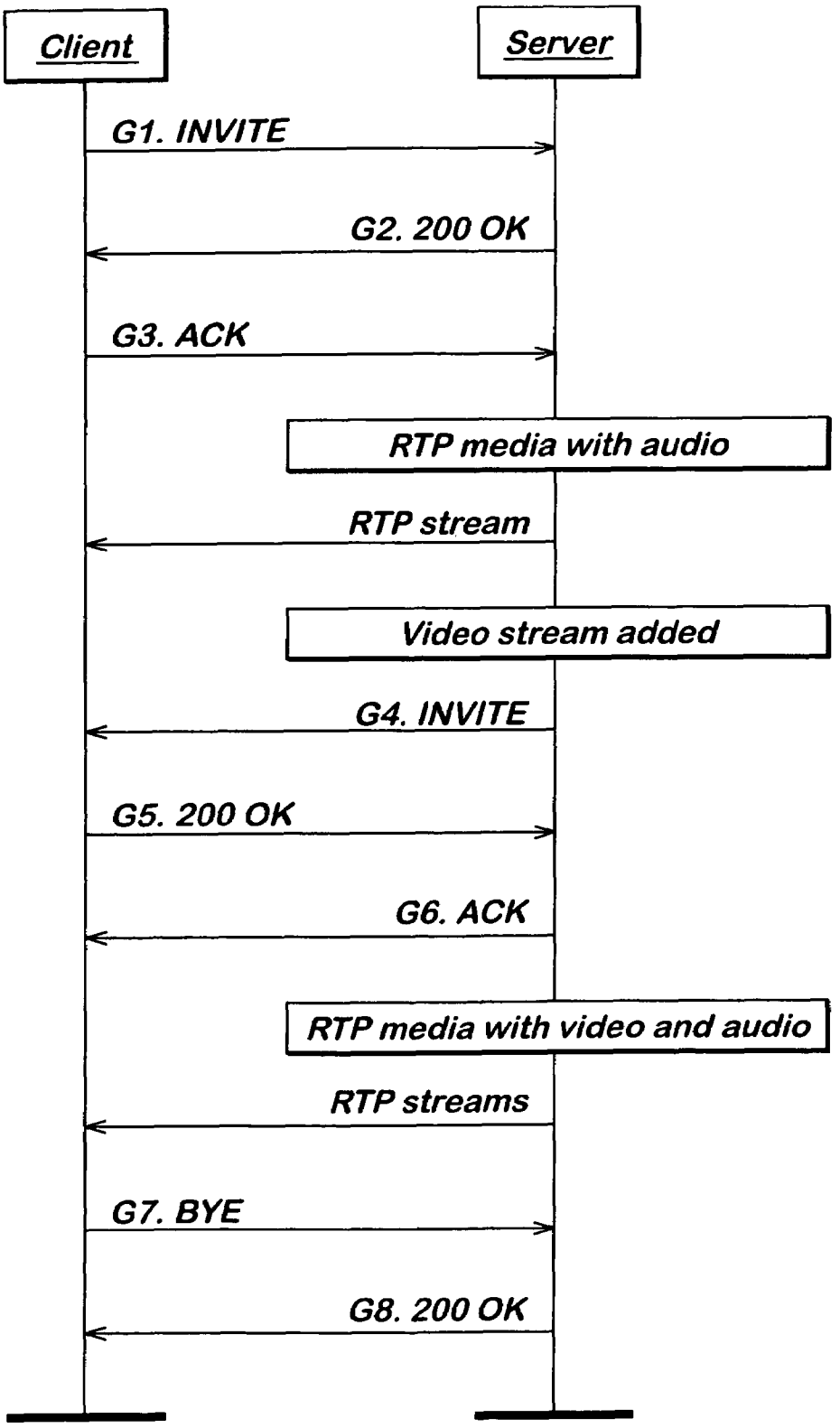

Client
Server
G1. INVITE
G2. 200 OK
G3. ACK
RTP media with audio
RTP stream
Video stream added
G4. INVITE
G5. 200 OK
G6. ACK
RTP media with video and audio
RTP streams
G7. BYE
G8. 200 OK

Fig. 8

Changes: Play -> setup, and additional steps 200 OK as H13 and H14 PLAY

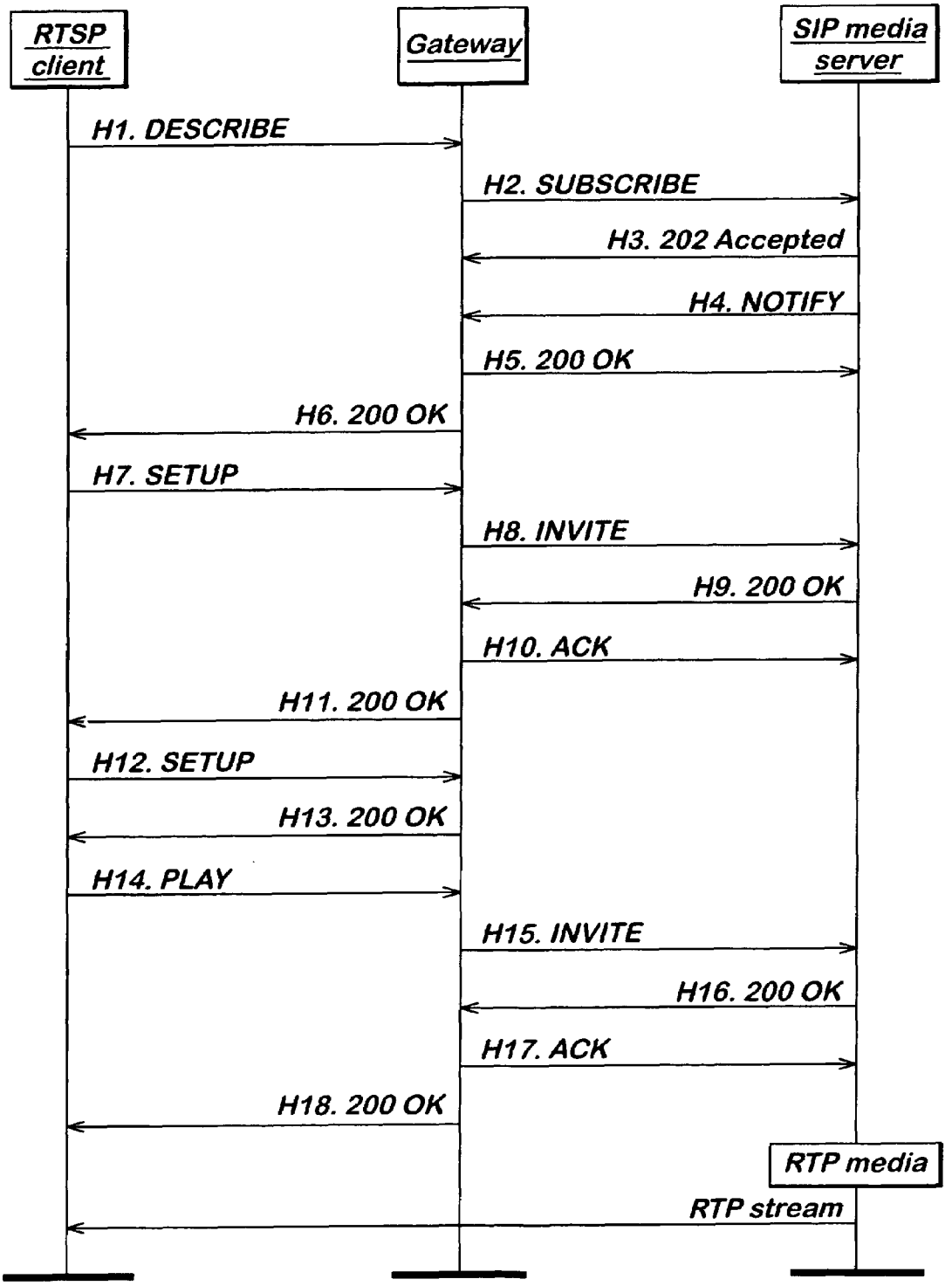
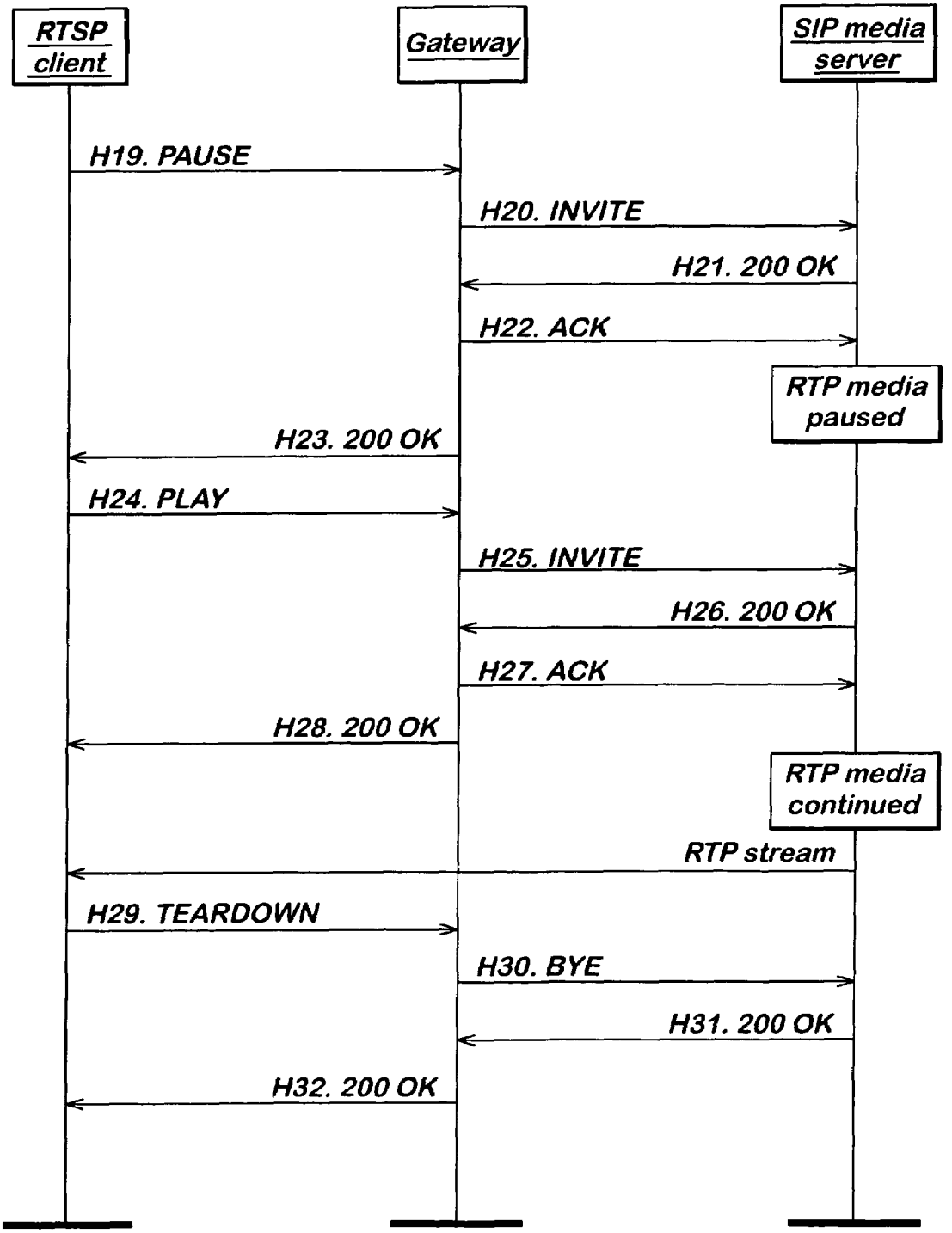


Fig. 9

With changes on message numbers (n+2), added new H12 and H13



STREAMING MEDIA

FIELD OF THE INVENTION

The present invention relates generally to streaming media.

BACKGROUND OF THE INVENTION

A communication system is a facility that enables communication between two or more entities such as user equipment and/or other nodes associated with the system. The communication may comprise, for example, communication of voice, data, multimedia and so on. Communication is shifting to communication networks such as IP (Internet Protocol) and/or other packet switched data networks. A communication system typically operates in accordance with a given standard or specification which sets out what the various elements of the system are permitted to do and how that should be achieved. For example, the standard or specification may define if the user, or more precisely, the user equipment or terminal is provided with a circuit switched service and/or a packet switched service. Communication protocols and/or parameters which shall be used for the connection may also be defined. In other words, a specific set of "rules" on which the communication can be based needs to be defined to enable communication by means of the system. The communication between the user equipment and the elements of the communication network can be based on an appropriate communication protocol, such as the SIP (Session initiated protocol), RTSP (real time streaming protocol) or some other protocol.

RTSP is a standard protocol for controlling multimedia streaming on Internet. The RTSP-protocol however has some disadvantages. In particular, RTSP does not provide for media negotiation and the client and server cannot dynamically select best possible media formats. Media negotiation was handled at application level, for instance, several RTSP URL to a particular media resource could be given, each with particular media parameters. The RTSP DESCRIBE method can be used to retrieve stream parameters from the RTSP streaming server. No negotiation is done and the streaming is possible only if the client supports the media and codecs listed in the response to DESCRIBE.

SIP (Session Invitation Protocol, RFC 3261) is a popular protocol used to setup multimedia sessions. Its applications include, for instance, Internet telephony and conferencing. SIP provides means for publishing endpoint capabilities, negotiating session contents (which media are used, which codecs are used, etc.) and updating sessions. It also includes means for subscribing to events and delivering event documents whenever the event status has been changed. It can handle negotiations for communications sessions dynamically. However, the SIP protocol does not support multimedia streaming.

SUMMARY OF THE INVENTION

It is an aim of embodiments of the present invention to address the above mentioned problems.

According to a first aspect of the present invention there is provided a method for providing an SIP session between a first and a second entity, said method comprising the steps of establishing a SIP session between the first and second entity; transmitting at least one media stream between the first entity and the second entity and controlling at least one of transmission, storage and play back of the at least one media stream at said first and/or second entity.

According to a second aspect of the present invention there is provided a method of providing an SIP session between a first and a second entity, and a RTSP session between said second entity and a third entity, said method comprising the steps of establishing a SIP session between the first and second entity; establishing an RTSP session between said second entity and said third mapping between SIP and RSTP information to allow data to pass from one of said first and third entities to the other of said first and third entities.

According to a third aspect of the present invention, there is provided a method of providing a SIP session between a first entity and a second entity, said method comprising the steps of establishing a SIP session between the first and second entities; providing at least one media stream to one of said first and second entities from the other of said entities; and updating said at least one media stream during said session.

According to a fourth aspect of the present invention, there is provided a communications system comprising a first and second entity, said first and second entity being operable to establish a SIP session between the first and second entity, one of said entities being arranged to transmit at least one media streams to the other entity, said system being such that at least one of transmission, storage and play back of the media streams is controlled at least one of said entities.

According to a fifth aspect of the present invention, there is provided a communications system comprising a first and second entity, said first and second entity being operable to establish a SIP session between the first and second entity, said first entity being arrange to retrieve a description of a session, said first and second entities being arranged to establish a session therebetween, one of the entities being arranged to send at least one media stream to the other entity and said other entity being arranged to control the play back of the media streams.

According to a sixth aspect of the present invention, there is provided a communications system comprising a first and a second entity, said first and second entity being operable to establish a SIP session between the first and second entity, one of said entities being arranged to transmit at least one media stream to the other entity, wherein said at least one of said media streams can be sent at one of a plurality of different playback rates.

According to a seventh aspect of the present invention, there is provide a communications system comprising a first, second and third entity wherein the first and second entities are arranged to establish a SIP session therebetween, and the second and third entities are arrange to establish a RTSP session therebetween, said second entity being arranged to map between SIP and RSTP information to allow data to pass from one of said first and third entities to the other of said first and third entities.

According to an eighth aspect of the present invention, there is provided a communications system comprising a first and a second entity, said first and second entity being operable to establish a SIP session between the first and second entity, one of said entities being arranged to transmit at least one media stream to the other entity, wherein said at least one media stream us updatable during said session.

According to a ninth aspect of the present invention, there is provided a gateway for use in a communications system, wherein said gateway is arranged to establish a SIP session with one entity and a RTSP session with a further entity, said gateway being arranged to map between SIP and RSTP infor-

mation to allow data to pass from one of said first and third entities to the other of said first and third entities.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding of the present invention and as to how the same may be carried into effect, reference will now be made by way of example to the accompanying drawings in which:

FIG. **1** illustrates streaming with SIP in which by using SIP-MEX (SIP with media streaming extensions) streaming media content is supported in SIP domain.

FIG. **2** illustrates a gateway between a streaming client and an RTSP-server which is used to enable streaming from RTSP to SIP domain.

FIG. **3** illustrates gateway between RTSP domain and SIP media server, where the stream is fetched from the SIP Media Server to RTSP domain.

FIG. **4** illustrates an example of SIP in streaming

FIG. **5** illustrates an example of voicemail with SIP

FIG. **6** illustrates updating of stream parameters in streaming

FIG. **7** illustrates an alternative updating of stream parameters in streaming

FIGS. **8** and **9** illustrates use of gateway between RTSP and SIP, where RTSP clients connects to a gateway and receives a video stream and RTSP messages are mapped to SIP messages. **7**

DETAILED DESCRIPTION OF THE INVENTION

FIG. **1** shows a schematic diagram of streaming with SIP, in accordance with embodiments of the invention. A streaming client A**20** is using a user application A**21** to receive streaming media from a user Application A**11** of a streaming server A**10**. SIP-MEX A**30** is used to provide streaming features to SIP.

FIG. **2** shows a schematic diagram of streaming media content between a RTSP Server B**11** and a SIP User Application B**21**, in accordance with an embodiment of the invention. The streaming server B**10** has an RTSP server B**11** which sends streaming media content via a RTSP connection B**31** to a RTSP Client B**41** of a gateway B**40**. The gateway B**40** handles the mapping of the streaming media originating from the RTSP server B**11** to a user application B**42** of the gateway so that it is compatible with the SIP-MEX B**30** connection. The user application B**42** of the gateway sends the streaming media content via the SIP-MEX B**30** connection to the user application B**21** of the streaming client B**20**, which is in the SIP domain.

FIG. **3** shows schematically an embodiment of the invention with a gateway C**40** between a RTSP domain C**20** and a SIP media server C**10**, where the stream is fetched from the SIP Media Server C**10** to RTSP domain C**20**. The user application C**11** of the streaming server C**10** sends streaming media content via a SIP-MEX connection C**31** to the user application C**41** of the gateway C**40**. The gateway C**40** handles the mapping of the streaming media originating from the user application of the streaming server C**11** to the RTSP server C**42** so that it is compatible with the RTSP connection C**30**. The RTSP Server C**42** sends the streaming media content via the RTSP connection C**30** to the RTSP Client C**21** of the streaming client C**20** which is in the RTSP domain.

FIG. **4** illustrates messaging between a client and a server, using SIP streaming. For the sake of clarity, features of the messages are explained only once, and previously explained or known SIP information is assumed to be understood by any person skilled in the art. In this example, the client requests streaming media, and decides that the movie or any other multimedia stream should be played. The user also decides to stop the streaming during the movie.

The communication is started by sending a SUBSCRIBE D**1** message from the client to the server. This message is used to indicate that a client wants to request streaming media.

The first message D**1** SUBSCRIBE from Client to Server is in the following form:

```
SUBSCRIBE sip:movies/matrix@media.nokia.com;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:movies/matrix@media.nokia.com;user=stream>
Cseq: 1 SUBSCRIBE
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Expires: 0
Event: session-description
Accept: application/sdp
Content-length: 0
```

This message is now explained in more detail:

SUBSCRIBE sip:movies/matrix@media.nokia.com; user=stream SIP/2.0

The method of the message is SUBSCRIBE. This field is used to describe what the client is requesting and that the version of SIP used is 2.0. Use of URI (uniform resource identifier) user part "movies/matrix" defining the source of requested streaming media is additional information proposed by embodiments of the invention compared to current features of SIP-protocol. Additionally, use of the URI parameter "user=stream", which defines that streaming is to be used, is also an additional feature proposed by embodiments of the present invention to the current SIP-protocol. These additional features to the SIP-protocol are referred as SIP-MEX.

From: <sip:user@nokia.com>

This header field is indicating from whom the message is originating. From field is typical field in the present SIP-protocol message and the value of the field is in accordance with present version of SIP-protocol. However use of SIP-MEX may require use of additional information in this field, so that address of the streaming media can be defined accurately.

To: <sip movies/matrix@media.nokia.com;user=stream>

This header field is indicating for whom the message is intended. To field is a typical field in the present SIP-protocol message, however in this case the field is additionally used in embodiments of the invention and contains additional information compared to present version of SIP identifying the source of the multimedia stream.

Cseq: 1 SUBSCRIBE

This header field indicates the sequence number of the sent request. Cseq is a feature of the present version of SIP.

Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95

This header field indicates the identity of this connection. Call-ID is a feature of the present version of SIP.

Contact. <sip.172.21.164.126>

This header field indicates the actual SIP host-port address of the client where the notifications from the other end will be sent. Contact is a feature of present version of SIP.

Expires: 0

This field header indicates the duration of the subscription. The expires value of 0 indicates that subscription is terminated immediately after initial notification is sent. In other words, the sender of SUBSCRIBE request only wants to

query the current description of streaming resource identified by the URI sip:movies/matrix@media.nokia.com. Expires is feature of present version of SIP.

Event: session-description

This header field indicates the event package to which the client is subscribing. Event names are used to separate different SIP event types from each other. Event header is a feature of present version of SIP, but the session-description event type is an additional feature provided by SIP-MEX.

Accept: application/sdp

This header field indicates that client will only accept responses containing the stream parameters in SDP (session description protocol) format. Accept is feature of present version of SIP.

Content-length: 0

This header field indicates the length of the message payload (number of characters). Content-length is feature of present version of SIP.

After receiving the SUBSCRIBE message D**1** from the Client, the Server answers the Client with message **202** Accepted D**2**. The response acknowledges that the other end has received the SUBSCRIBE message and the client can start waiting for NOTIFY messages.

The message **202** Accepted from the server to the client is as follows:

```
SIP/2.0 202 Accepted
To: <sip:movies/matrix@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:movies/matrix@131.228.54.140;user=stream>
Event: session-description
Cseq: 1 SUBSCRIBE
Content-length: 0
```

SIP/2.0 **202** Accepted

The status code of this response message is **202** with reason phrase "Accepted". This field indicates the class of the response message.

Contact: <sip:movies/matrix@131.228.54.140;user=stream>

This header field indicates the source of the streaming media. In this case the field is only compatible with SIP-MEX because it is contains the additional information discussed previously, as compared to present version of SIP.

All other fields of this message are similar to those previously explained.

The server also sends to the client a NOTIFY D**3** message. The server encloses the stream parameters in the payload of the NOTIFY message and can push the information to the client dynamically.

The message D**3** NOTIFY from the server to the client:

```
NOTIFY sip:172.21.164.126 SIP/2.0
From: <sip:movies/matrix@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:movies/matrix@131.228.54.140;user=stream>
CSeq: 1 NOTIFY
Event: session-description
Range: smtpe-25=00:00:00–02:16:13
Subscription-State: terminated;reason=timeout
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 2 IN IP4 131.228.54.140
```

-continued

```
s=The Matrix
u=http://us.imdb.com/Title?0133093
t=0 0
a=control: sip:movies/matrix@media.nokia.com;user=stream
a=send only
m=audio 0 RTP/AVP 96 97 98
a=rtpmap:96 AMR-WB/16000
a=rtpmap:97 AMR/8000
a=rtpmap:98 L16/48000/2
m=video 0 RTP/AVP 26 31
```

The method of this message is NOTIFY, its request-URI is sip:172.21.164.126. The field indicates the method name and the address where the message is sent.

Range: smtpe-25=00:00:00-02:16:13

This header field indicates the length of the movie or multimedia stream. Range is an additional feature provided by SIP-MEX, in embodiments of the invention.

Subscription-State: terminated;reason=timeout

This header field indicates the state of the SIP event subscription. The value "terminated" indicates that the subscription is no more active. The "reason" parameter indicates reason for termination. In this case, the one-shot subscription is terminated with the first notification.

Content-type: application/sdp

This header field indicates the MIME multimedia internet mail extension type of the content.

Content-length header field indicates the length of the content and is a feature of the present version of SIP.

The rest of the message belongs to its contents, which consists of stream description in SDP language.

v=0

The version "v=" field indicates the version of SDP in use. Field v is feature of present version of SIP.

o=media-server 1804289383 2 IN IP4 131.228.54.140

The origin "o=" field indicates the identity and version of the session description. Field o is feature of present version of SIP.

s=The Matrix

The subject "s=" field indicates the subject of the stream. Field s is feature of present version of SIP.

u=http://us.imdb.com/Title?0133093

This field indicates the URL where additional information about the session can be obtained.

t=0 0

The time "t=" field indicates the start and stop time of the multicast sessions. In this case, the start and stop time depend on client action, and the field is included only because SDP syntax requires it. Field t is feature of the present version of SIP.

The attribute "a=" fields contain optional information about session or media. Field a is a feature of the present versions of SIP and SDP. The SIP user agent will ignore the SDP attributes it does not know.

a=control: sip:movies/matrix@media.nokia.com;user=stream

The attribute "control" indicates the URI that can be used to manipulate the media stream. The "control" attribute is only compatible with SIP-MEX because it contains additional information compared to present version of SIP.

a=sendonly

The send only attribute field indicates that media session is only unidirectional, from server towards the client. The "send only" attribute is a feature of the present version of SIP.

m=audio 0 RTP/AVP 96 97 98

The "m=" media field indicates the type of a media stream within the streaming session. It contains the media type (audio), the transport port (0) and protocol (RTP/AVP) and then alternative media formats (expressed as RTP payload numbers). The server is able to send media using any of the media formats listed here. The client can choose the media format from the list provided by the server based on its own preferences and the network capacity available to it. Field m is feature of present version of SIP

a=rtpmap:96 AMR-WB/16000

The "rtpmap" attribute indicates mapping from RTP payload number 96 to an audio format AMR-WB with sample rate of 16 kHz. The "rtpmap" attribute is feature of present version of SIP.

a=rtpmap:97 AMR/8000

The "rtpmap" attribute indicates mapping from RTP payload number 97 to an audio format AMR with sample rate of 8 kHz. The "rtpmap" attribute is feature of present version of SIP.

a=rtpmap:98 L16/48000/2

The "rtpmap" attribute indicates mapping from RTP payload number 98 to an audio format L16 with sample rate of 48 kHz and two channels. The "rtpmap" attribute is feature of present version of SIP.

m=video 0 RTP/AVP 26 31

The "m=" media field indicates the media type of a stream within the session. It contains the media type (video), the transport port (0) and protocol (RTP/AVP) and then alternative media formats (expressed as well-known RTP payload numbers, 26 stands for JPEG, 31 for H.261).

All other fields of this message are similar to those previously explained.

After the client receives the NOTIFY (D**3**) message from the server, the Client sends a 200 OK message D**4** to the server to confirm that it has received parameters for the streaming connection.

The message 200 OK from Client to Server is as follows.

```
SIP/2.0 200 OK
To: <sip:user@nokia.com>
From: <sip:movies/matrix@media.nokia.com;user=stream>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
CSeq: 1 NOTIFY
Content-length: 0
```

The status code of this response message is 200 with reason phrase "OK". The status code 200 indicates success.

All other fields of this message are similar to those previously explained.

The user interface on the client can now show the stream information on the user terminal. The client starts streaming the movie. The user wants to start fast rewind with the picture starting from the end of the movie, so the Range and Scale Headers are sent in the INVITE message D**5**.

The message INVITE from Client to Server is as follows

```
INVITE sip:movies/matrix@media.nokia.com;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:movies/matrix@media.nokia.com;user=stream>
Cseq: 1 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Range: smtpe-25=-02:16:13
Scale: -2.0
Content-type: application/sdp
Content-length: 131
v=0
```

-continued

```
o=-816357754 1 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
m=audio 20582 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
m=video 20584 RTP/AVP 31
```

The method of this request message is INVITE with the request URI sip:movies/matrix@media.nokia.com; user=stream SIP/2.0. The field indicates the name of the SIP method and the SIP address of the recipient. The user part (movies/matrix) in the address specifies the stream at the server. The parameter user=stream provides the network elements with an option to separate streaming handling from VoIP (voice over IP) calls.

Scale: −2.0

This field indicates that the playback rate of the stream is set to 2 (that is twice the normal rate) and that the direction of the stream is reversed due to the negative value. Scale is an additional feature provided by SIP-MEX in embodiments of the present invention

c=IN IP4 172.21.164.126

The connection "c" field indicates the network address of the client. The client asks server to send the media streams to the address specified in field c. Field c is feature of present version of SIP.

a=recvonly

The "recvonly" attribute indicates that client wants to establish an unidirectional media session, from server towards the client. The "recvonly" attribute is a feature of the present version of SIP.

m=audio 20582 RTP/AVP 96

a=rtpmap:96 AMR-WB/16000

The media description, consisting of field m and the rtpmap attribute, indicates the type of media stream within the streaming session that the client is prepared to receive. It contains the media type (audio), the transport port (20582) and transport protocol (RTP/AVP) that the client uses to receive media. After the transport protocol, the client then lists the media format (expressed as dynamic RTP payload number on the "m=" line and the "rtpmap" attribute below it) that it expects to receive. The server is able to use any of the media formats listed here to send the media stream. If the streaming resource does not contain media in the listed formats, the server can either convert the format in streaming resource to the format accepted by client, or if it cannot do converting, decline the request. The session description in E**3** contained the formats readily supported by server, so the client should include one of the formats listed in E**3** in its request here. Field m is feature of present version of SIP.

All other fields of this message are similar to those explained previously.

After receiving the INVITE message D**5** from the client, the server sends a confirmation message 200 OK D**6** to the client.

The message 200 OK from the server to the client is also follows:

```
SIP/2.0 200 OK
To: <sip:movies/matrix@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
```

-continued

```
Cseq: 1 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:movies/matrix@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
Range: smtpe-25=00:00:00–02:16:13
Scale: −1.80
Speed: 1
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=The Matrix
a=control: sip:movies/matrix@131.228.54.140;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=sendonly
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
m=video 32850 RTP/AVP 26
```

The status code of this response message is 200 with reason phrase "OK". The status code 200 indicates success.

Speed: 1

This field value indicates that no buffering is to be used, the server will send stream at the same rate as it should be played out by the client. Speed is an additional feature provided by SIP-MEX in embodiments of the present invention.

All other fields of this message similar to those described previously.

After the client receives confirmation message 200 OK D**6** from the server, the client acknowledges the server that it has received 200 OK D**6** message by sending ACK message D**7** to the server.

The message ACK from the client to the server is as follows

```
ACK sip:movies/matrix@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:movies/matrix@media.nokia.com;user=stream>
Cseq: 1 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The method of this message is ACK.

All other fields of this message are similar to those discussed earlier.

The server starts RTP media streaming. When the client wants to terminate the stream, it sends a BYE message D**8** to the server.

The message BYE from the client to the server is as follows

```
BYE sip:movies/matrix@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:movies/matrix@media.nokia.com;user=stream>
Cseq: 2 BYE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The method of this request message is BYE, its request-URI is sip:movies/matrix@131.228.54.140;user=stream. The purpose of this message is to terminate the session.

All other fields of this message similar to those discussed previously.

When the server receives from the client the message BYE it sends a confirmation message 200 OK D**9** to the client and stops the streaming.

The message 200 OK from the server to the client is as follows

```
SIP/2.0 200 OK
To: <sip:movies/matrix@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Cseq: 2 BYE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The status code of this response message is 200 with reason phrase "OK". The status code 200 indicates success.

All other fields of this message are similar to those already described.

In a second example of using streaming media in SIP, this example involves using and controlling a voice mail box with SIP. FIG. **5** illustrates how messaging between parties involved are exchanged. Individual messages and actions between the parties are explained after each message. For the sake of clarity, only new features in the messages are explained. The first party, Alice, tries to call the second party, Bob, but the incoming call is redirected to an voicemail server. Alice hears a pre-recorded message from the voicemail server. Alice decides to leave a voicemail message for Bob in the voice mailbox. After recording a voicemail message Alice wants to review the message she is leaving in the voice mailbox. The voicemail server plays to Alice the message she has left for Bob and then Alice tears down the connection with the voicemail server. In this use case there are included streaming control parameters (Range, Scale, Speed) in the SDP instead of using SIP headers.

Communication is started by sending an INVITE message E**1** from the client to the server. This message is used to advise that a client wants to request streaming media.

The message INVITE from Alice to Bob takes the following form:

```
INVITE sip:bob@nokia.com SIP/2.0
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 1 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=−816357754 1 IN IP4 172.21.164.126
s=−
t=0 0
c=IN IP4 172.21.164.126
m=audio 20582 RTP/AVP 96 8 0
a=rtpmap:96 AMR/8000
```

The fields are as already described.

When Bob receives the INVITE message E**1** from Alice, the request for connection is answered with message **302** Moved Temporarily E**2**

The message Moved Temporarily from Bob to Alice E**2** is as follows

```
SIP/2.0 302 Moved Temporarily
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 1 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
```

-continued

```
Contact: <sip:bob@mailbox.nokia.com;user=stream>
Content-Type: text/html
Content-Length: 121
<html>
<title>Please leave a message</title>
<body>
Please leave a message at <a href="sip:bob@mailbox.nokia.com">
my voicemail</a>.
</body>
</html>
```

Topic of this message is SIP/2.0 **302** Moved Temporarily, which indicates effectively that Bob is unavailable and diverts the call to the voice mail box.

In the Hypertext Markup Language (HTML) part of this message data is transferred from Bob to Alice. Alice will receive a request to leave message to Bob's voicemail and a hyperlink to the voice mailbox is included in the message. The other fields are as already described.

Alice responds to the **302** Moved Temporarily message from Bob with message E**3**—ACK to acknowledge that she have received message from Bob.

The message ACK from Alice to Bob is as follows:

```
ACK sip:bob@nokia.com SIP/2.0
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 1 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-Length: 0
```

The fields are as already described.

Alice can now redirect her connection to the voicemail server. Alice sends an INVITE message E**4** to the voicemail server.

The message INVITE form Alice to Server is as follows:

```
INVITE sip:bob@mailbox.nokia.com;user=stream SIP/2.0
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 2INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=−816357754 1 IN IP4 172.21.164.126
s=−
t=0 0
c=IN IP4 172.21.164.126
m=audio 20582 RTP/AVP 96 8 0
a=rtpmap:96 AMR/8000
```

The topic of this message is INVITE sip:bob@mailbox.nokia.com;user=stream SIP/2.0. The URI parameter user=stream indicates that streaming media will be used. The header and SDP fields are as already described.

After receiving the INVITE message from Alice, the server confirms that it has received the message from Alice and the Server send a 200 OK message E**5** to Alice.

The 200 OK message from the server to Alice is as follows:

```
SIP/2.0 200 OK
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
```

-continued

```
Cseq: 2 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:bob@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
a=control:sip:bob@131.228.54.140;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=sendonly
a=range:ntp=−32.0
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR/8000
```

a=control:sip:bob@131.228.54.140;user=stream

The control attribute field indicates the URI that is used to control the streaming media resource. The control attribute field is only compatible with SIP-MEX because it contains additional information compared to present version of SIP, but it will be silently ignored by any SIP user-agent not supporting SIP-MEX.

a=range:ntp=−32.0

The range attribute field indicates the length of the pre-recorded message. The range attribute field is only compatible with SIP-MEX because it contains additional information compared to present version of SIP, but it will be silently ignored by any SIP user-agent not supporting SIP-MEX.

The other fields are as already described.

After Alice receives the 200 OK message from the server, Alice responds to the Server with the message ACK E**6** to inform the server that Alice is ready to receive streaming media from the Server.

The message ACK from Alice to the Server is as follows:

```
ACK sip:bob@131.228.54.140;user=stream SIP/2.00
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 2 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The message E**6** ACK is similar to message E**3** ACK. In E**6** ACK Alice informs the server that streaming will be used.

After the Server receives message E**6** ACK from Alice, the Server starts playing out the pre-recorded message form Bob, which lasts 32 seconds as indicated in message 200 OK by the Server. After the pre-recorded message is played, Alice is given a chance to record her 60-second voicemail to Bob. Alice is sent an INVITE message E**10** updating the session and she is redirected to a URL specific to her message at the same time, by using a contact header.

The message INVITE E**7** sent from the server to Alice is as follows:

```
INVITE sip:172.21.164.126 SIP/2.0
From: <sip:bob@nokia.com>
To: <sip:alice@nokia.com>
Cseq: 0 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:bob/msg.10101@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
```

-continued

```
o=media-server 1804289383 321764 IN IP4 131.228.54.140
s=-
t=0 0
c=IN IP4 131.228.54.140
a=control:sip:bob/msg.10101@131.228.54.140;user=stream
a=sendrecv
a=range:ntp=0.0–60.0
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR/8000
```

Contact: <sip:bob/msg.10101@131.228.54.140; user=stream>

This field indicates that the streaming media connection should be directed to the given address. In this case, the field Contact has additional feature provided by SIP-MEX.

a=control:sip:bob/msg.10101@131.228.54.140; user=stream

The control attribute field indicates the URI that is used to control the streaming media resource. The attribute is different from the URI given in earlier SDP, and the change indicates that the media resource to which Alice is now being redirected is different from resource played out earlier.

a=range:ntp=0.0-60.0

The range attribute field indicates that the length of the message Alice can record for Bob is something between 0 seconds and 60 seconds. The range attribute field is an additional feature, provided by embodiments of the invention.

The other fields are as already described.

Alice then decides to leave a voicemail in to the Bob's voice mailbox. Alice sends an 200 OK message E**8** to the server and confirms that she wants to leave a voicemail.

The message 200 OK from Alice to the server is as follows:

```
SIP/2.0 200 OK
From: <sip:bob@nokia.com>
To: <sip:alice@nokia.com>
Cseq: 0 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=-816357754 2 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
m=audio 20582 RTP/AVP 96
a=rtpmap:96 AMR/8000
```

The fields are as already described.

After the server receives the confirmation message E**8** from Alice, the server acknowledges to Alice that it has received message E**8** by sending an acknowledgment ACK message E**9** to Alice.

The message E**9** from the server to Alice is as follows:

```
ACK sip:172.21.164.126 SIP/2.0
From: <sip:bob@nokia.com>
To: <sip:alice@nokia.com>
Cseq: 0 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The fields are as already described.

Alice has now a streaming connection with the server, and Alice can now record her message to Bob. However, Alice wants to review her message, which she can do it by sending an INVITE message E**10** to the server. Note that the Request URI is the URL provided in the Contact header of the message E**10** sent by the voicemail server. The same URL is also specified in the updated SDP. The Request URI is specific to the voicemail message recorded by Alice.

The message E**10** from Alice to the Server is as follows

```
INVITE sip:bob/msg.10101@mailbox.nokia.com;user=stream SIP/2.0
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 3 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=-816357754 3 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
a=range:ntp=0.0-
a=recvonly
m=audio 20582 RTP/AVP 96 8 0
a=rtpmap:96 AMR/8000
```

The fields are as already described.

After receiving message E**10** from Alice, the Server answers Alice with a 200 OK message E**11** confirming that it has received the message from Alice.

The message E**11** from the server to Alice is as follows

```
SIP/2.0 200 OK
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 3 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:bob/msg.10101@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321764 IN IP4 131.228.54.140
s=-
t=0 0
c=IN IP4 131.228.54.140
a=control:sip:bob/msg.10101@131.228.54.140;user=stream
a=sendonly
a=range:ntp=0.0–27.6
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR/8000
```

The fields are as already described.

After Alice receives the 200 OK message from the Server, Alice responds to the Server with an ACK message E**12** to inform the server that Alice is ready to listen to the message that she has left for Bob, so that Alice receives the streaming media from the server

The message E**12** from Alice to the Server is as follows

```
ACK sip:bob/msg.10101@131.228.54.140;user=stream SIP/2.0
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 3 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The fields are as already described.

After Alice has reviewed her voicemail to Bob, Alice decides to disconnect from the server, so she hangs up the call. The message E**13**—BYE is sent from Alice to the server.

The message E**13** BYE from Alice to the server is as follows:

```
BYE sip:bob/msg.10101@131.228.54.140;user=stream SIP/2.0
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 4 BYE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The fields are as already described.

The server receives message E**13** from Alice and it responds to that message by sending message E**14**—200 OK to Alice to confirm the teardown of the connection.

The message E**14** from the server to Alice is as follows:

```
SIP/2.0 200 OK
From: <sip:alice@nokia.com>
To: <sip:bob@nokia.com>
Cseq: 4 BYE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The fields are as already described

In the third example of using streaming media in SIP, this case presents an advanced case of use of SIP in streaming. FIG. **6** illustrates messaging between a client and a server. In this case a client requests streaming a live media stream (audio) from the server. While the original stream (audio) is playing, a stream (video) is added to it. Changes in the stream are made with NOTIFY message which notifies the Client about a new stream (video). The client decides to receive the new stream (video), too. The Client negotiates the media again, now with video support. Messaging happens simultaneously, so the original stream (audio) need not be cancelled.

Communication between the Client and Server is started by the message F**1** SUBSRCIBE from Client to Server. Steps from F**1** to F**7** are the similar to steps D**1** to D**7** of FIG. **4**. The full messages for messages F**1** to F**7** are set out below.

Message F1

```
SUBSCRIBE sip:reports/interim/q2-2002@media.nokia.com;
user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Cseq: 1 SUBSCRIBE
Call-ID: 4df6e7d8-5834-11d6-a882-00e00307ab95
Contact: <sip:172.21.164.126>
Expires: 3600
Event: session-description
Accept: application/sdp
Content-length: 0
```

Message F2

```
SIP/2.0 202 Accepted
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 4df6e7d8-5834-11d6-a882-00e00307ab95
Cseq: 1 SUBSCRIBE
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Event: session-description
Content-length: 0
```

-continued

Message F3

```
NOTIFY sip:172.21.164.126 SIP/2.0
From: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 4df6e7d8-5834-11d6-a882-00e00307ab95
Cseq: 1 NOTIFY
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Event: session-description
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 2 IN IP4 131.228.54.140
s=Nokia Interim Report Q2/2002
t=0 0
a=control: sip:reports/interim/q2-2002@media.nokia.com;user=stream
a=sendonly
m=audio 0 RTP/AVP 96 97 98
a=rtpmap:96 AMR-WB/16000
a=rtpmap:97 AMR/8000
a=rtpmap:98 L16/48000/2
```

Message F4

```
SIP/2.0 200 OK
To: <sip:user@nokia.com>
From: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 4df6e7d8-5834-11d6-a882-00e00307ab95
Cseq: 1 NOTIFY
Content-length: 0
```

Message F5

```
INVITE sip:reports/interim/q2-2002@media.nokia.com;
user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 3 INVITE
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=- 816357754 1 IN IP4 172.21.164.126
s=-.
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
m=audio 20582 RTP/AVP 96 98
a=rtpmap:96 AMR-WB/16000
a=rtpmap:97 L16/16000/2
```

Message F6

```
SIP/2.0 200 OK
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 4 INVITE
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Nokia Interim Report Q2/2002
t=0 0
c=IN IP4 131.228.54.140
a=control: sip:reports/interim/q2-2002@media.nokia.com;user=stream
a=sendonly
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
```

Message F7

```
ACK sip:reports/interim/q2-2002@media.nokia.com;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 4 ACK
Content-length: 0
```

The connection between the client and the server is now established, and RTP media streaming starts. Next, the ses-

sion description is updated within the media server and the media server notifies the client about a video stream that can be added to the session. The client asks server to add the video stream to the original stream, this is done in a similar way to the way in which the original audio streaming was started. Steps from F**8** to F**12** are similar to steps F**3** to F**7**, so these steps are not described in detail. These messages are set out below.

Message F8

```
NOTIFY sip:172.21.164.126 SIP/2.0
From: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 4df6e7d8-5834-11d6-a882-00e00307ab95
Cseq: 5 NOTIFY
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Event: session-description
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 3 IN IP4 131.228.54.140
s=Nokia Interim Report Q2/2002
t=0 0
a=control: sip:reports/interim/q2-2002@media.nokia.com;user=stream
a=sendonly
m=audio 0 RTP/AVP 96 97 98
a=rtpmap:96 AMR-WB/16000
a=rtpmap:97 AMR/8000
a=rtpmap:98 L16/48000/2
m=video 0 RTP/AVP 26 31
```

Message F9

```
SIP/2.0 200 OK
To: <sip:user@nokia.com>
From: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 4df6e7d8-5834-11d6-a882-00e00307ab95
Cseq: 5 NOTIFY
Content-length: 0
```

Message F10

```
INVITE sip:reports/interim/q2-2002@131.228.54.140:user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 6 INVITE
Contact: <sip:172.21.164.126>
Accept: application/sdp
Range: npt=0.000-
Content-type: application/sdp
Content-length: 131
v=0
o=- 816357754 1 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
m=audio 20582 RTP/AVP 96 98
a=rtpmap:96 AMR-WB/16000
a=rtpmap:97 L16/16000/2
m=video 20584 RTP/AVP 26 31
```

Message F11

```
SIP/2.0 200 OK
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 6 INVITE
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Nokia Interim Report Q2/2002
a=control: sip:reports/interim/q2-2002@media.nokia.com;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=sendonly
```

-continued

```
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
m=video 32850 RTP/AVP 26
```

Message F12

```
ACK sip:reports/interim/q2-2002@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 6 ACK
Content-length: 0
```

RTP Media streaming continues now with the both audio and video streams. When the Client wants to teardown the connection, the client sends message F**13** BYE to the server. Steps F**13** and F**14** are typical connection teardown messages and they are similar to steps D**8** and D**9** of FIG. **4**. These messages are set out below:

Message F**13**

Message F13

```
BYE sip:reports/interim/q2-2002@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 7 BYE
Content-length: 0
```

Message F14

```
SIP/2.0 200 OK
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 7 BYE
Content-length: 131
```

An alternative way to update stream parameters is shown in FIG. **7**, which illustrates messaging between a client and a server. In this example updating streaming content is done by a server-initiated INVITE message. The new media stream (video) is negotiated and added simultaneously while the audio stream is played.

Communication between the Client and Server is started by the message G**1** INVITE from Client to Server. Steps from G**1** to G**3** are similar to steps E**4** to E**6**, steps G**4** to G**6** are similar to steps E**7** to E**9** and steps G**7** and G**8** are similar to steps E**13** and E**14** of FIG. **4**. The messages are set out below.

Message G1

```
INVITE sip:reports/interim/q2-2002@media.nokia.com;
user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 1 INVITE
Contact: <sip:172.21.164.126>
Accept: application/sdp
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=- 816357754 1 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
```

-continued

```
m=audio 20582 RTP/AVP 96 98
a=rtpmap:96 AMR-WB/16000
a=rtpmap:97 L16/16000/2
m=video 20584 RTP/AVP 26 31
```

Message G2

```
SIP/2.0 200 OK
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 1 INVITE
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Nokia Interim Report Q2/2002
t=0 0
c=IN IP4 131.228.54.140
a=control: sip:reports/interim/q2-2002@media.nokia.com;user=stream
a=sendonly
m=audio 32852 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
m=video 0 RTP/AVP 26 31
```

Message G3

```
ACK sip:reports/interim/q2-2002@media.nokia.com;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 1 ACK
Content-length: 0
RTP Media streaming starts with audio only.
```

Message G4

```
INVITE sip:172.21.164.126 SIP/2.0
From: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 1 INVITE
Contact: <sip:reports/interim/q2-2002@131.228.54.140;user=stream>
Accept: application/sdp
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321764 IN IP4 131.228.54.140
s=Nokia Interim Report Q2/2002
t=0 0
c=IN IP4 131.228.54.140
a=control: sip:reports/interim/q2-2002@media.nokia.com;user=stream
a=sendonly
m=audio 33400 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
m=video 33402 RTP/AVP 26 31
```

Message G5

```
SIP/2.0 200 OK
From: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 1 INVITE
Contact: <sip:172.21.164.126>
Content-type: application/sdp
Content-length: 131
v=0
o=- 816357754 2 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
m=audio 20582 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
m=video 20584 RTP/AVP 26
```

Message G6

```
ACK sip:172.21.164.126 SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
```

-continued

```
Cseq: 1 ACK
Content-length: 0
RTP Media streaming continues now with the both audio and video.
When the Client wants to teardown the connection, normal teardown
procedure is done.
```

Message G7

```
BYE sip:reports/interim/q2-2002@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 3 BYE
Content-length: 0
```

Message G8

```
SIP/2.0 200 OK
To: <sip:reports/interim/q2-2002@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Cseq: 3 BYE
Content-length: 0
```

The last example describes an ordinary RTSP client connecting to a gateway and it initiates receiving a video stream. FIGS. **8** and **9** illustrate the messaging between RTSP client, gateway and SIP media server, with FIG. **8** showing the first part of the signaling and FIG. **9** showing the second part of the signaling.

The request for streaming media by the RTSP client is announced by message H**1** DESCRIBE from the RTSP client to the gateway.

Message H1

```
0 DESCRIBE rtsp://media-gw.nokia.com/twister RTSP/1.0
CSeq: 1
```

This message is a RTSP message and it indicates what the RTSP client is requesting and where from, also version of RTSP is indicated to be 1.0

When the gateway receives message H**1**, it maps the message so that it can create a SIP-MEX message from it. The gateway sends message H**2**—SUBSCRIBE to the SIP media server. Communication between the gateway and SIP Media Server is similar to previous examples, so that steps from H**2** to H**5** are similar to steps D**1** to D**4** in FIG. **4**. These messages are:

Message H2

```
SUBSCRIBE sip:twister@media.nokia.com;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 1 SUBSCRIBE
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Expires: 0
Event: session-description
Accept: application/sdp
Content-length: 0
```

Message H3

```
SIP/2.0 202 Accepted
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
```

-continued

```
Event: session-description
Cseq: 1 SUBSCRIBE
Content-length: 0
```

Message H4

```
NOTIFY sip:131.228.237.22 SIP/2.0
From: <sip:twister@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
CSeq: 1 NOTIFY
Event: session-description
Range: smtpe-25=00:00:00–02:16:13
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 2 IN IP4 131.228.54.140
s=Twister
a=control: sip:twister@media.nokia.com;user=stream
t=0 0
a=sendonly
m=audio 0 RTP/AVP 0
m=video 0 RTP/AVP 26
```

Message H5

```
SIP/2.0 200 OK
To: <sip:user@nokia.com>
From: <sip:twister@media.nokia.com;user=stream>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
CSeq: 1 NOTIFY
Content-length: 0
```

The gateway sends information about the requested streaming media to the RTSP client in message H6 —200 OK.

Message H6

```
RTSP/1.0 200 OK
CSeq: 1
Content-Type: application/sdp
Content-Length: 164
v=0
o=- 2890844256 2890842807 IN IP4 172.16.2.93
s=Twister
a=control:rtsp://media-gw.nokia.com/twister
t=0 0
m=audio 0 RTP/AVP 0
a=control:audio
m=video 0 RTP/AVP 26
a=control:video
```

Fields in this message are common to RTSP and will be clear to a person skilled in the art.

The RTSP Client now starts streaming the movie and sends message H**7**—SETUP to the gateway.

Message H7

```
SETUP rtsp://media-gw.nokia.com/twister/audio RTSP/1.0
CSeq: 2
Transport: RTP/AVP;unicast;client_port=8000–8001
```

Fields in this message are common to RTSP.

The gateway receives the H**7** message from RTSP client and interprets message H**7** to a form that the SIP media server can understand. The gateway sends message H**8**—INVITE to the SIP media server. Communication between the gateway and the SIP media server is similar to previous examples, so that steps from H**8** to H**10** are similar to steps D**5** to D**7** in FIG. **4**.

Message H8

```
INVITE sip:twister@media.nokia.com;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 1 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Content-length: 0
```

Message H9

```
SIP/2.0 200 OK
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Cseq: 1 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Twister
a=control: sip:twister@131.228.54.140;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=sendonly
a=range:smtpe-25=00:00:00–02:16:13
m=audio 9004 RTP/AVP 0
a=rtpinfo: seq=92781365;rtptime=21375134
m=video 9002 RTP/AVP 26
a=rtpinfo: seq=9810092;rtptime=3450012
```

Message H10

```
ACK sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 1 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 83
v=0
o=– 816357754 1 IN IP4 172.21.164.126
s=–
t=0 0
c=IN IP4 172.21.164.126
a=inactive
m=audio 8000 RTP/AVP 0
m=video 0 RTP/AVP 26
```

The gateway sends confirmation to the RTSP client in message H**11**—200 OK.

Message H11

```
RTSP/1.0 200 OK
CSeq: 2
Transport: RTP/AVP;unicast;client_port=8000–8001;
    source=131.228.54.140;server_port=9004–9005
Session: 12345678
```

Fields in this message are common to RTSP.

The RTSP client sets up also the video session and sends message H**12** SETUP to the gateway

Message H12

```
SETUP rtsp://media-gw.nokia.com/twister/video RTSP/1.0
CSeq: 3
Transport: RTP/AVP;unicast;client_port=8002–8003
```

Fields in this message are common to RTSP.

The Gateway confirms the request H**12** for adding of the video session by sending message H**13**—200 OK to the RTSP client.

Message H13

```
RTSP/1.0 200 OK
CSeq: 3
Transport: RTP/AVP;unicast;client_port=8002–8003;
    source=131.228.54.140;server_port=9002–9003
Session: 12345678
```

Fields in this message are common to RTSP.

The RTSP client requests that playing of the streaming media starts and sends message H**14** to the gateway.

Message H14

```
PLAY rtsp://media-gw.nokia.com/twister RTSP/1.0
CSeq: 4
Range: npt=0–
Session: 12345678
```

The gateway requests to the SIP media server that video streaming should also be included. The gateway sends message H**15**—INVITE to the SIP media server. Communication between the gateway and the SIP media server is similar to the previous examples, so that steps H**15** to H**17** are similar to steps H**8** to H**10**.

Message H15

```
INVITE sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 2 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Content-length: 0
v=0
o=– 816357754 2 IN IP4 172.21.164.126
s=–
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
a=range:ntp=0–
m=audio 8000 RTP/AVP 0
m=video 8002 RTP/AVP 26
```

Message H16

```
SIP/2.0 200 OK
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Cseq: 2 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Twister
a=control: sip:twister@131.228.54.140;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=sendonly
a=range:ntp=0–
m=audio 9000 RTP/AVP 0
a=rtpinfo: seq=656823807;rtptime=2829945
m=video 9002 RTP/AVP 26
a=rtpinfo: seq=9810092;rtptime=3450012
```

-continued

Message H17

```
ACK sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 2 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 83
```

The gateway sends confirmation to the RTSP client in message H**18**—200 OK.

Message H18

```
RTSP/1.0 200 OK
CSeq: 4
Session: 12345678
RTP-Info: url=rtsp://media-gw.nokia.com/twister/audio
    ;seq=656823807;rtptime=2829945
RTP-Info: url=rtsp: //media-gw.nokia.com/twister/video
    ;seq=9810092;rtptime=3450012
```

Fields in this message are common to RTSP.

The RTSP Client now receives the streaming media from the SIP Media Server. Next the RTSP Client wants to pause the streaming media from the SIP Media Server. The RTSP Client sends message H**19**—Pause to the Gateway.

Message H19

```
PAUSE rtsp://media-gw.nokia.com/twister RTSP/1.0
CSeq: 5
Session: 12345678
```

Fields in this message are common to RTSP.

Gateway receives this message H**19** PAUSE and it interprets it to a form that the SIP Media Player can understand it. Gateway sends message H**20** to the SIP media server to pause the streaming. Messages H**20** and H**21** have value "inactive" in their field a of the message—this feature of embodiments of the present invention pauses the streaming in the SIP Media Server. Otherwise steps H**20** to H**22** similar to steps H**8** to H**10** earlier in this example.

Message H20

```
INVITE sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 3 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Content-length: 0
v=0
o=– 816357754 2 IN IP4 172.21.164.126
s=–
t=0 0
c=IN IP4 172.21.164.126
a=inactive
m=audio 8000 RTP/AVP 0
m=video 8002 RTP/AVP 26
```

Message H21

```
SIP/2.0 200 OK
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
```

-continued

```
Cseq: 3 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Twister
a=control: sip:twister@131.228.54.140;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=inactive
m=audio 9000 RTP/AVP 0
m=video 9002 RTP/AVP 26
```

Message H22

```
ACK sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 3 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 83
```

RTP media is now paused. The gateway sends confirmation to the RTSP client in message H**23**—200 OK.

Message H23

```
RTSP/1.0 200 OK
CSeq: 5
Session: 12345678
```

Fields in this message are common to RTSP.

The RTSP client is ready to resume playing the stream, so the Client sends message H**24**—PLAY to the Gateway.

Message H**24** PLAY from RTSP Client to Gateway

```
PLAY rtsp://media-gw.nokia.com/twister RTSP/1.0
CSeq: 6
Range: npt=15-
Session: 12345678
```

Fields in this message are common to RTSP.

The gateway requests that the SIP media server resume the streaming media from the point it was paused. The gateway sends message H**25**—INVITE to the SIP Media Server. Communication between the gateway and the SIP media server is similar to previous steps, that is steps H**25** to H**27** are similar to steps H**13** to H**15**.

Message H25

```
INVITE sip:twister@ 131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 4 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Content-length: 0
v=0
o=- 816357754 2 IN IP4 172.21.164.126
s=-
t=0 0
c=IN IP4 172.21.164.126
a=recvonly
a=range: npt=15-
m=audio 8000 RTP/AVP 0
m=video 8002 RTP/AVP 26
```

-continued

Message H26

```
SIP/2.0 200 OK
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Cseq: 4 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@ 131.228.54.140;user=stream>
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 321763 IN IP4 131.228.54.140
s=Twister
a=control: sip:twister@ 131.228.54.140;user=stream
t=0 0
c=IN IP4 131.228.54.140
a=sendonly
a=range: npt=15-
m=audio 9000 RTP/AVP 0
a=rtpinfo:seq=656823882;rtptime=2949945
m=video 9002 RTP/AVP 26
a=rtpinfo:seq=9810467;rtptime=3450027
```

Message H27

```
ACK sip:twister@ 131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 4 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 83
```

The gateway sends confirmation to the RTSP client in message H**28**—200 OK.

Message H28

```
RTSP/1.0 200 OK
CSeq: 6
Session: 12345678
RTP-Info: url=rtsp://media-gw.nokia.com/twister/audio
    ;seq=656823882;rtptime=2949945
RTP-Info: url=rtsp://media-gw.nokia.com/twister/video
    ;seq=9810467;rtptime=3450027
```

Fields in this message are common to RTSP.

The RTSP Client again receives streaming media from the SIP media server. Next the RTSP client wants to stop the streaming media from the SIP media server. The RTSP Client sends message H**29**—TEARDOWN to the Gateway.

Message H29

```
TEARDOWN rtsp://media-gw.nokia.com/twister RTSP/1.0
CSeq: 7
Session: 12345678
```

Fields in this message are common to RTSP.

The gateway receives message H**29**—TEARDOWN from the RTSP client. The typical teardown procedure is done. Message H**30**—BYE is sent to the SIP media server to start the teardown. Steps H**30** and H **31** are similar to steps D**8** and D**9** of FIG. **4**.

Message H30

```
BYE sip:twister@ 131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
```

-continued

```
Cseq: 5 BYE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

Message H31

```
SIP/2.0 200 OK
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Cseq: 5 BYE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 0
```

The Gateway sends confirmation to the RTSP client in message H**32**—200 OK.

Message H32

```
RTSP/1.0 200 OK
CSeq: 7
```

Fields in this message are common to RTSP.

In embodiments of the invention, a communication network is able to dynamically select best format for streaming media regardless whether the domain of the user is based on RTSP or SIP. In embodiments of the invention, media streaming extensions for the SIP protocol are proposed in order to create a version of SIP with media streaming extensions. In the embodiments, this is SIP-MEX, which adds all the RTSP functionality to SIP. Embodiments of the invention enable mapping between RTSP and SIP-MEX, and connection between SIP and RTSP-domains can be achieved with a gateway between them.

With SIP-MEX there is no need for a separate streaming protocol or network proxies for streaming services. According to one embodiment of the invention, existing SIP architecture can be used to provide an easy billing scheme to streaming services, and user applications can access streaming media more easily. The streaming media resources, like video clips, can be accessed with a plain Internet telephone application.

In accordance with another embodiment of the present invention, in internet telephony SIP requests to streaming media resources can be distinguished from telephony requests, and the streaming media can be cached.

By adding the functionality of multimedia streaming SIP, it is possible to use media streaming parameters (range, scale, speed) in the SIP messages. In embodiments of the invention streaming media format may be selected dynamically in RTSP environment, which means updating media session during the streaming. On the other hand that SIP environment can support RTSP-functionality, which enables mapping between RTSP and SIP-MEX, and their: URLs (Uniform Resource Locator).

Embodiments of the invention provide a to establish a media session in SIP/SDP with properly negotiated medias. The SIP protocol or the SDP (Session Description Protocol) descriptions used by SIP is extended so that they can provide all the RTSP functionality. The SIP with media streaming extensions (SIP-MEX) can be used in various network configurations as described above.

Thus embodiments of the invention may provide a method for mapping between RTSP URLs and SIP URLs. The URL to the media resource is given in RTSP request line, e.g. SETUP rtsp://media.nokia.com/movies/matrix RTSP/1.0 When a SIP element is given a media resource URL, e.g. rtsp://media.nokia.com/movies/matrix, it needs to convert the RTSP URL to a SIP URL. The mapping is straightforward: the host:port part is left as is and the SIP username part is used to express the media resource, in this case, movies/matrix, like as follows: INVITE sip:movies/matrix@media.nokia.com SIP/2.0 To distinguish between the media stream resources and legacy SIP usernames, like sip:user@nokia.com, and make it possible to unambiguously map between SIP and RTSP URLs, the parameter user=stream can be used: INVITE sip:movies/matrix@media.nokia.com;user=stream SIP/2.0

The user=stream parameter tells the Sims registrar, proxy or application server (AS) to handle the requests to media streaming resources properly (i.e., do not mix them with normal users). A proxy can also cache streaming media accessed with a user=stream parameter. An UA processing an SIP URL with user=stream parameter can add the streaming controls (play, rewind, pause, etc.) to the user interface.

Embodiments of the invention may provides means for updating media session during the streaming. SIP event extension, SUBSCRIBE—NOTIFY methods, can be used to retrieve the stream parameter data expressed in Session Description Protocol (SDP). If the stream parameters change during the playback, e.g. a video channel is added to a live stream, a new NOTIFY containing updated SDP can be sent to the client. The client then can send a new INVITE (re-INVITE) to the server and media is negotiated again.

When using RTSP, the Range header specifies the point where the playback starts. No such header currently exists in SIP. There is two alternative solutions, the Range header can be added to SIP or a Range attribute can be added to SDP.

The syntax of the RTSP Range header is as follows:

Range="Range""=" 1\#ranges-specifier

["=" "time" "=" utc-time]

ranges-specifier=npt-range|utc-range|smpte-range

Example as a SIP Header:

Range: clock=19960213T143205Z-; time=19970123T143720Z

Example as an SDP Attribute:

a=range:clock=19960213T143205Z-; time=19970123T143720Z

Scale header is used in the RTSP playback or record to view/listen the media stream at different playback rates (implementing fast forward/fast rewind). For example, with Scale: 3.0 the server would transmit media with timestamps changing three times as fast as normally. Just like Range, SIP needs to be extended with Scale header or a Scale attribute could be added to the SDP description.

Example as a SIP Header:

Scale: −1.0

Example as an SDP Attribute:

a=scale:−1.0

Speed is used to ask the server to deliver playback stream at a particular transfer rate, for example, with Speed: 3.0 the server would transmit media with playback time of 3.0 seconds within 1.0 second. The playback rate is not affected by transfer rate. This allows client to buffer the stream and avoid interrupts in the playback (due to network congestion). Again, a new SIP header or SDP attribute can be used.

Consider the following example of a video stream. It contains 25 frames a second. The server puts a timestamp to each frame, and client plays the frames at the time indicated by the timestamp.
The stream is normally transmitted like this:
A..B..C..D..E..F..G..H..I..J..K..L..M..N..O..P..Q..R..S..T..U..V..W..X..Y..Z
And played like this
A..B..C..D..E..F..G..H..I..J..K..L..M..N..O..P..Q..R..S..T..U..V..W..X..Y..Z
In this case, the real time, the time in video clip and time indicated by timestamps go at the same rate.
With Speed: 3.0, the frames are transmitted like this:
ABCDEFGHIJKLMNOPQRSTUVWXYZabcdefghijklmnopqrstuvwxyz0123456789
@!#$%&/ ©=+?- ®¤
But played out like this:
A..B..C..D..E..F..G..H..I..J..K..L..M..N..O..P..Q..R..S..T..U..V..W..X..Y..Z
In this case, the time indicated by timestamps (and video time) go three times faster than the real time. However, the playout happens at normal rate.
With Scale: 3.0, they are transmitted like this (assuming the streaming server is intelligent enough to drop extra frames away):
A..D..G..J..M..P..S..V..Y..b..e..h..k..n..q..t..w..z..2..5..8..!..%.. ©..?..¤
And played out like this:
A..D..G..J..M..P..S..V..Y..b..e..h..k..n..q..t..w..z..2..5..8..!..%.. ©..?..¤
In this case, the real time and time indicated by timestamps go again at the same rate, but the time in video clip is accelerated.

SIP event extension, SUBSCRIBE—NOTIFY methods, can be used to retrieve the stream parameter data expressed in Session Description Protocol (SDP). If the stream parameters change during the playback, e.g., a video channel is added to a live stream, a new NOTIFY containing updated SDP can be sent to the client. The client then can send a new INVITE (re-iNVITE) to the server and media is negotiated again.

Parameter update can also be handled with re-INVITE, sent by the streaming server. There is no need to use NOTIFY to bear change information. The re-INVITE is now used in opposite direction, the streaming server sends the re-INVITE and the streaming client receives it. Note, however, that a RTSP-SIP gateway can not always honor the re-INVITE sent by streaming server.

Parameter update can also be handled with re-INVITE, sent by the streaming server. There is no need to use NOTIFY to bear change information. The re-INVITE is now used in opposite direction, the streaming server sends the re-INVITE and the streaming client receives it.

It should be appreciated that is some embodiments of the invention, the initially established SIP session may be without streaming with the streaming being added and/or negotiated later. In other embodiments of the invention, the SIP session maybe established with a media stream session.

The implementation is mostly done in application level. The protocol level (SIP/SDP) implementation is by adding a support for extension headers/attributes. The application level implementation is done for the server and client.

The invention claimed is:

1. A method for receiving a media stream at a first device from a second device using a session invitation protocol session, the method comprising:

sending a first session invitation protocol (SIP) message from a first device to a second device, wherein the first SIP message includes a first indicator indicating a request for a media stream;

receiving a second SIP message from the second device at the first device, wherein the second SIP message includes a second indicator indicating acceptance of the request for the media stream;

sending a third SIP message from the first device to the second device, wherein the third SIP message includes a third indicator indicating that the second device initiate streaming of the requested media stream;

receiving the requested media stream from the second device at the first device;

receiving a fourth SIP message from the second device at the first device, wherein the fourth SIP message includes stream parameters for the requested media stream;

sending a sixth SIP message from the first device to the second device, wherein the sixth SIP message includes second stream parameters for the requested media stream, the second stream parameters defining parameters for the received media stream;

receiving a seventh SIP message from the second device at the first device, wherein the seventh SIP message includes an update to the stream parameters;

sending an eighth SIP message from the first device to the second device, wherein the eighth SIP message includes third stream parameters for the requested media stream defined based on the updated stream parameters; and

receiving the requested media stream from the second device at the first device based on the third stream parameters.

2. The method of claim **1**, wherein the first SIP message is a SUBSCRIBE message or an INVITE message.

3. The method of claim **1**, wherein the first SIP message further includes an identification of the source of the requested media stream.

4. The method of claim **3**, wherein the identification of the source of the requested media stream is a uniform resource identifier.

5. The method of claim **4**, wherein a parameter of the uniform resource identifier includes the first indicator.

6. The method of claim **1**, wherein the second SIP message further includes an identification of the source of the requested media stream.

7. The method of claim **1**, wherein the stream parameters include a length of the requested media stream.

8. The method of claim **1**, wherein the stream parameters include a playback rate for the requested media stream.

9. The method of claim **1**, wherein the stream parameters include a transfer rate for the requested media stream.

10. The method of claim **1**, wherein the stream parameters are included in a header field of the fourth SIP message.

11. The method of claim **1**, wherein the stream parameters are included in a payload of the fourth SIP message.

12. The method of claim **1**, wherein the stream parameters include a stream description.

13. The method of claim 12, wherein the stream description is defined in a session description protocol language.

14. The method of claim 12, wherein the stream description includes a media type and a media format supported by the second device.

15. The method of claim 1, further comprising sending a fifth SIP message from the first device to the second device, wherein the fifth SIP message includes a fourth indicator indicating receipt of the stream parameters.

16. The method of claim 1, wherein the second stream parameters include a length of the requested media stream.

17. The method of claim 1, wherein the second stream parameters include a playback rate for the requested media stream.

18. The method of claim 1, wherein the second stream parameters include a transfer rate for the requested media stream.

19. The method of claim 1, wherein the second stream parameters include a media format for the requested media stream, the media format selected from the stream parameters.

20. The method of claim 1, further comprising receiving a ninth SIP message from the second device at the first device, wherein the ninth SIP message includes a fifth indicator indicating confirmation of receipt of the second stream parameters.

21. The method of claim 1, further comprising sending a fourth SIP message from the first device to the second device, wherein the fourth SIP message includes a fourth indicator indicating that the second device stop streaming the requested media stream.

22. The method of claim 1, wherein the seventh SIP message is a NOTIFY message or an INVITE message.

23. The method of claim 1, wherein the second device is a gateway that processes real time streaming protocol (RTSP) messages from a third device and maps the RTSP messages to SIP messages.

24. A method for streaming a media stream between devices using a session invitation protocol session, the method comprising:

receiving a first real time streaming protocol (RTSP) message from a first device at a second device, wherein the first RTSP message includes a first indicator indicating a request for a media stream;

mapping the first RTSP message to a first session invitation protocol (SIP) message at the second device, wherein the first SIP message includes a second indicator indicating the request for the media stream;

sending the first SIP message from the second device to a third device;

receiving a second SIP message from the third device at the second device, wherein the second SIP message includes stream parameters for the requested media stream;

mapping the second SIP message to a second RTSP message, wherein the second RTSP message includes the stream parameters for the requested media stream;

sending the second RTSP message from the second device to the first device;

receiving a third RTSP message from the first device at the second device, wherein the third RTSP message includes second stream parameters for the requested media stream, the second stream parameters defining parameters for streaming the requested media stream;

mapping the third RTSP message to a third SIP message at the second device, wherein the third SIP message includes the second stream parameters;

sending the third SIP message from the second device to the third device;

receiving a fourth SIP message from the third device at the second device, wherein the fourth SIP message includes a third indicator indicating acceptance of the second stream parameters;

mapping the fourth SIP message to a fourth RTSP message, wherein the fourth RTSP message includes a fourth indicator indicating acceptance of the second stream parameters;

sending the fourth RTSP message from the second device to the first device;

receiving the requested media stream from the third device at the second device; and

streaming the received media stream from the second device to the first device.

25. The method of claim 24, further comprising:

receiving a fifth RTSP message from the first device at the second device, wherein the fifth RTSP message includes a fifth indicator indicating that the second device initiate streaming of the requested media stream;

mapping the fifth RTSP message to a fifth SIP message at the second device, wherein the fifth SIP message includes a sixth indicator indicating that the third device initiate streaming of the requested media stream; and

sending the fifth SIP message from the second device to the third device.

26. The method of claim 25, further comprising:

receiving a sixth RTSP message from the first device at the second device, wherein the sixth RTSP message includes a seventh indicator indicating that the second device stop streaming the requested media stream;

mapping the sixth RTSP message to a sixth SW message, wherein the sixth SIP message includes an eighth indicator indicating that the third device stop streaming the requested media stream;

sending the sixth SIP message from the second device to the third device.

27. A device for streaming a media stream between devices using a session invitation protocol session, the device configured to:

receive a first real time streaming protocol (RTSP) message from a second device, wherein the first RTSP message includes a first indicator indicating a request for a media stream;

map the first RTSP message to a first session invitation protocol (SIP) message, wherein the first SIP message includes a second indicator indicating the request for the media stream;

send the first SIP message to a third device;

receive a second SIP message from the third device, wherein the second SIP message includes stream parameters for the requested media stream;

map the second SIP message to a second RTSP message, wherein the second RTSP message includes the stream parameters for the requested media stream;

send the second RTSP message to the second device;

receive a third RTSP message from the second device, wherein the third RTSP message includes second stream parameters for the requested media stream, the second stream parameters defining parameters for streaming the requested media stream;

map the third RTSP message to a third SIP message, wherein the third SIP message includes the second stream parameters;

send the third SIP message to the third device;

receive a fourth SIP message from the third device, wherein the fourth SIP message includes a third indicator indicating acceptance of the second stream parameters;

map the fourth SIP message to a fourth RTSP message, wherein the fourth RTSP message includes a fourth indicator indicating acceptance of the second stream parameters;

send the fourth RTSP message to the second device;

receive the requested media stream from the third device; and

stream the received media stream to the second device.

* * * * *