



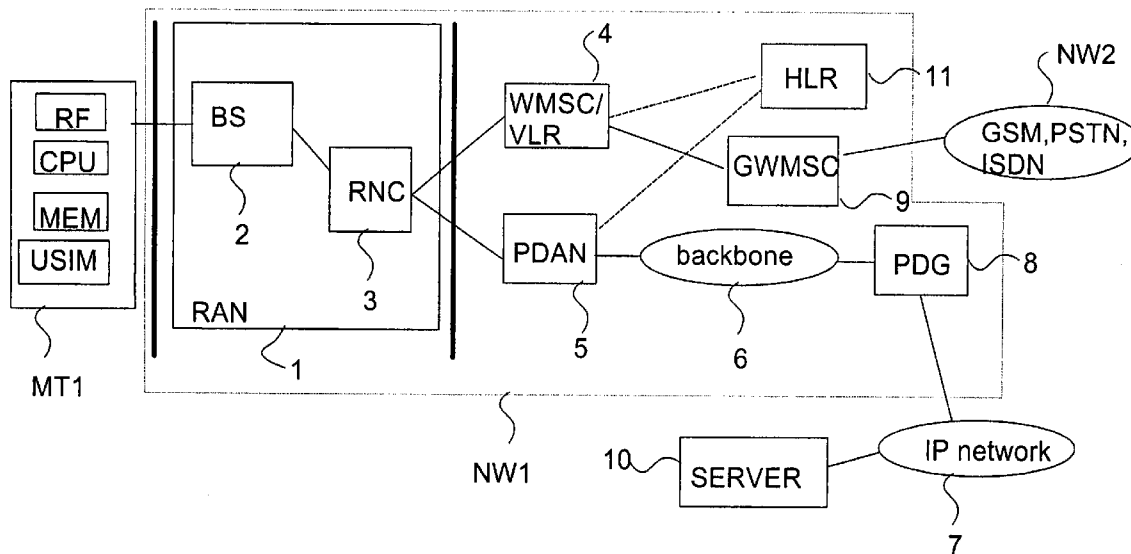
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(19) **United States**(12) **Patent Application Publication**
Curcio et al.(10) **Pub. No.: US 2005/0232148 A1**(43) **Pub. Date: Oct. 20, 2005**(54) **METHOD IN A COMMUNICATION SYSTEM,
A COMMUNICATION SYSTEM AND A
COMMUNICATION DEVICE**(60) Provisional application No. 60/550,443, filed on Mar.
4, 2004.(75) Inventors: **Igor D.D. Curcio**, Tampere (FI); **Emre
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MONROE, CT 06468 (US)**(57) **ABSTRACT**

The present invention relates to a method in a communication system, a communication system and a communication device. In the method, media streams are transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network. At least one media stream is selected to be transmitted to the receiving communication device, QoS requirements for transmitting the selected at least one media stream are defined, transmission resources are reserved from the wireless communication network for the transmission of the at least one media stream, and a setup procedure is performed between the receiving communication device and the sending communication device for activating the one packet data transmission connection.

(73) Assignee: **Nokia Corporation**(21) Appl. No.: **11/073,029**(22) Filed: **Mar. 3, 2005****Related U.S. Application Data**(63) Continuation-in-part of application No. 10/876,262,
filed on Jun. 24, 2004.

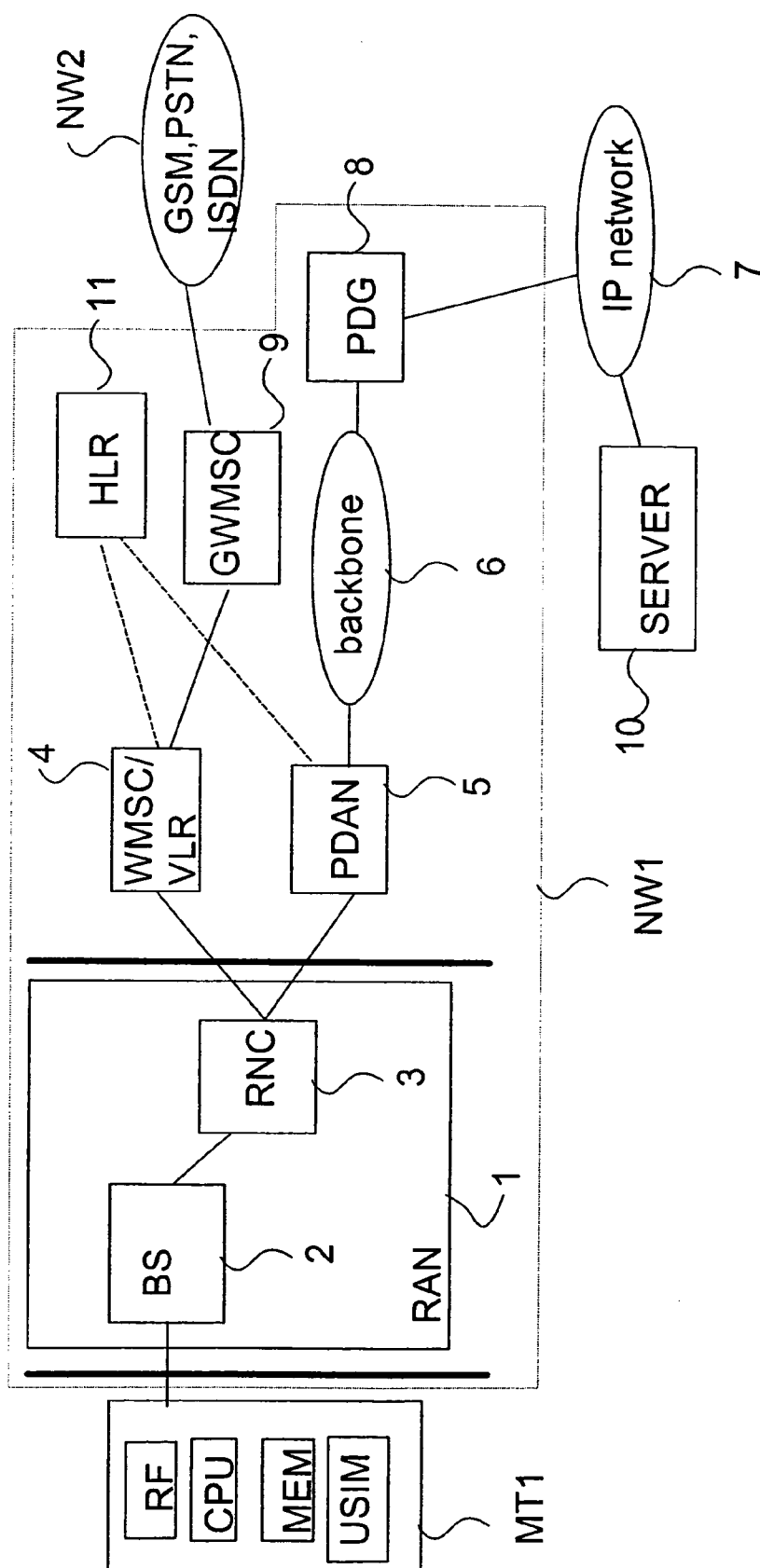


Fig. 1

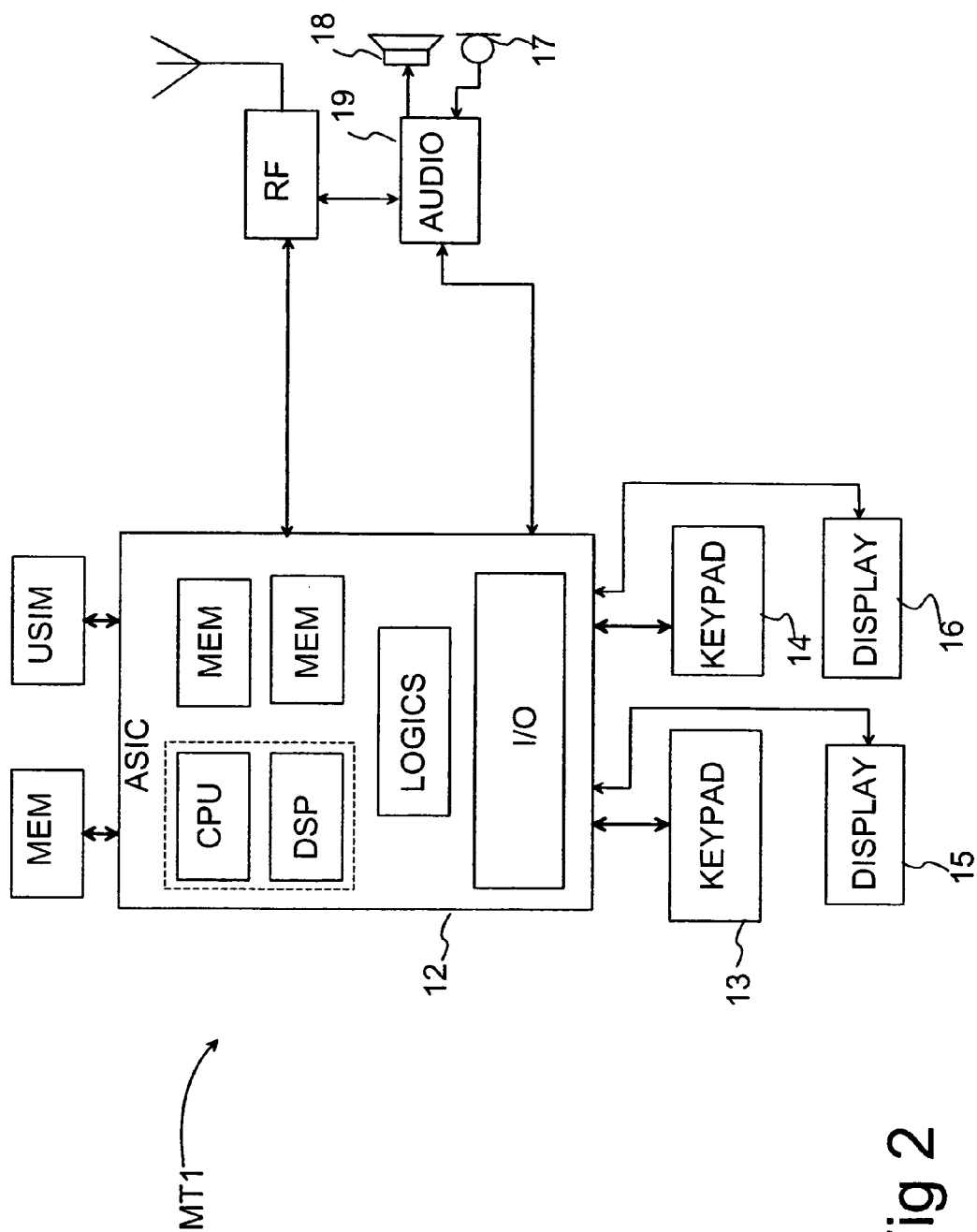


Fig 2

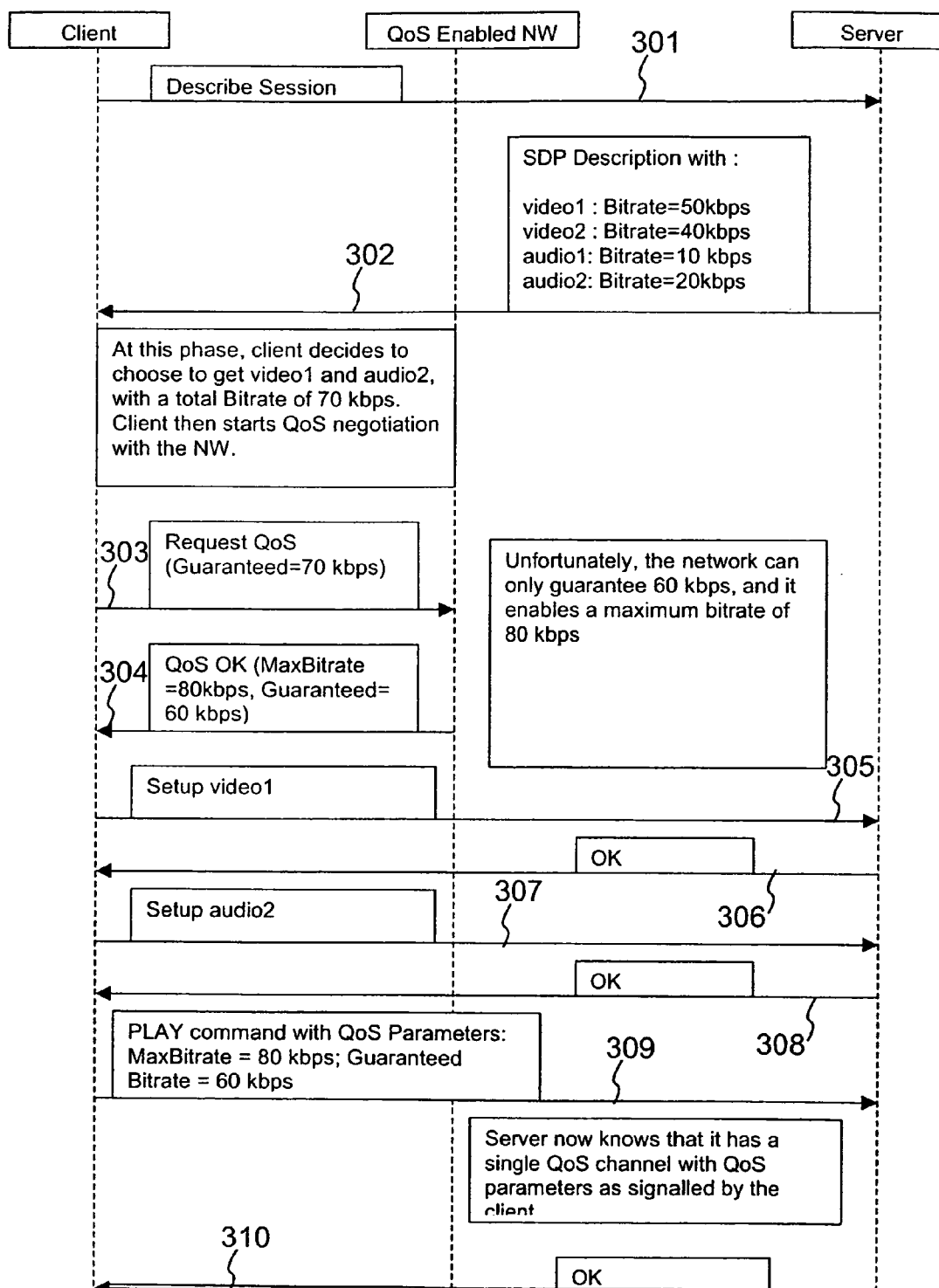
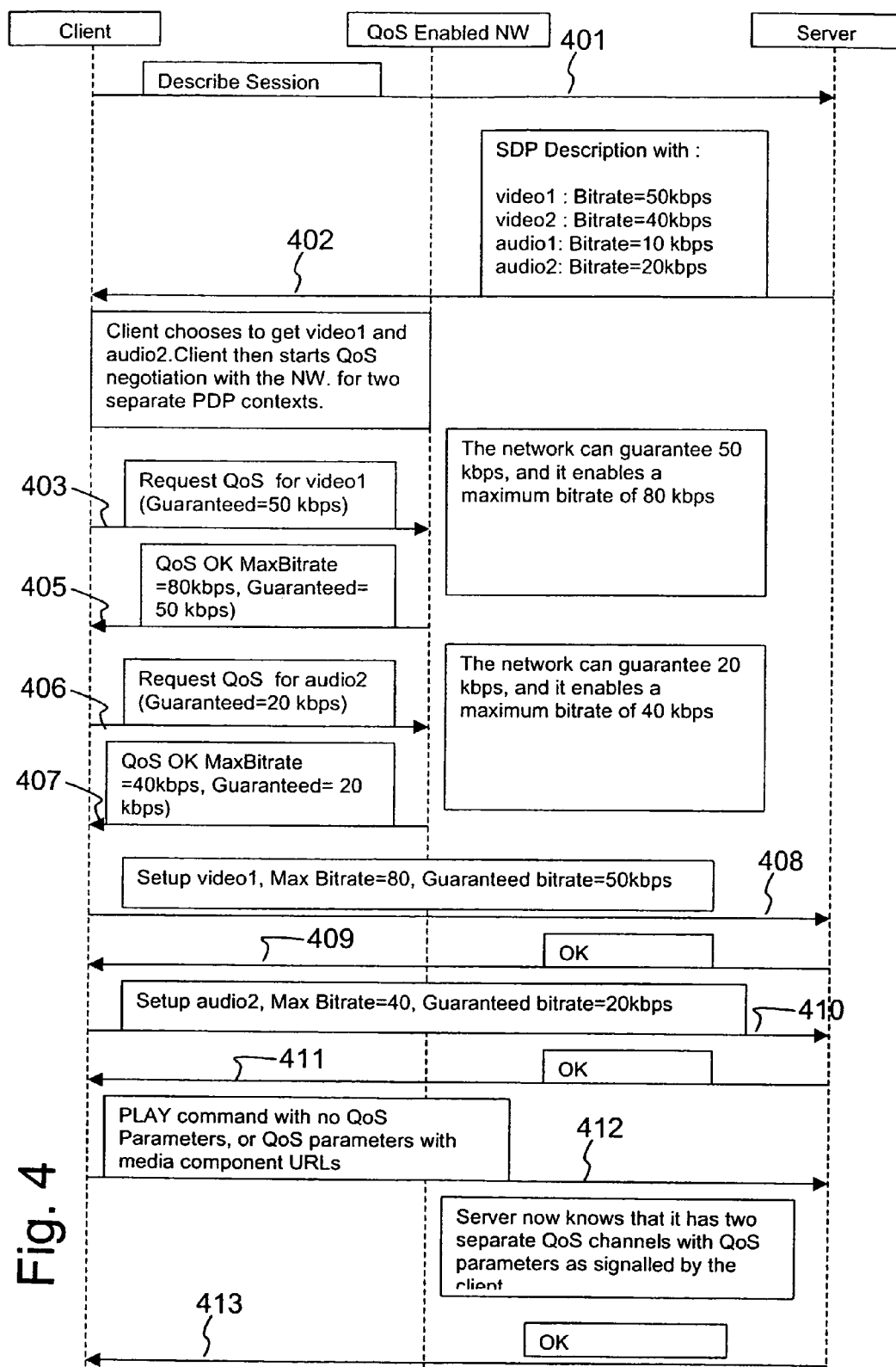


Fig. 3



METHOD IN A COMMUNICATION SYSTEM, A COMMUNICATION SYSTEM AND A COMMUNICATION DEVICE

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application claims priority under 35 USC §119 to U.S. Provisional Patent Application No. 60/550, 443 filed on Mar. 4, 2004 and also claims domestic priority under 35 USC §120 to U.S. patent application Ser. No. 10/876,262 filed Jun. 24, 2004.

FIELD OF THE INVENTION

[0002] The technology area is that of streaming media over mobile networks, where a multimedia server, a mobile network and a streaming client are logically connected e.g. via an RTSP protocol (Real Time Streaming Protocol) used for session setup and control, and e.g. an RTP protocol (Real Time transport Protocol) for media transfer. Streaming systems can be rate adaptive or not. This invention is related to rate adaptive streaming systems that can adapt the content and/or transmission rate to the varying network channel conditions.

[0003] The present invention relates to a method in a communication system, in which multimedia streams are transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network. The invention also relates to a communication system comprising a sending communication device, a receiving communication device, and a communication network to transmit multimedia streams from the sending communication device to the receiving communication device at least partly via a wireless communication network. The invention further relates to a sending communication device and a receiving communication device.

BACKGROUND OF THE INVENTION

[0004] In this description, the term sending communication device refers to a communication device including a transmitter being arranged to send multimedia streams to a communication network. The term receiving communication device refers to a communication device including a receiver for receiving multimedia streams from the communication network, respectively. It is obvious that the same communication device may include both the transmitter and the receiver whereby allowing one-way or two-way communication with the communication network. A wireless communication device includes a transmitter and/or a receiver implementing wireless communication in a wireless communication network. The term wireless communication system, such as a mobile communication system, generally refers to any communication system which makes a wireless data transmission connection possible between a wireless communication device and stationary parts of the system, the user of the wireless communication device moving within the operating range of the system. A typical wireless communication system is a Public Land Mobile Network (PLMN). A well-known example is the GSM system (Global System for Mobile telecommunications). The invention preferably relates to the third generation of mobile communication systems. As an example, the Universal Mobile Telecommunications System (UMTS) is used as an example of such a third-generation communication system.

[0005] In third generation systems, the terms bearer service and service are used. A bearer service is a telecommunication service type which provides the facility to transmit signals between access points. In general, the bearer service corresponds to the term of a traffic channel which defines, for example, the data transmission rate and the quality of service (QoS) to be used in the system when information is transmitted between a wireless communication device and another part of the system. The bearer service between the wireless communication device and the base station is, for example, a radio bearer service, and the bearer service between the radio network control unit and the core network is, for example, an lu bearer service (Interface UMTS bearer). In the UMTS system, the interface between the radio network control unit and the core network is called lu interface. In UMTS there is also the so called GERAN part, which uses, in addition to the lu interface, also an interface called as Gb interface. In this connection, the service is provided by the mobile communication network for performing a task (tasks); for example, data services perform data transmission in the communication system, telephone services are related to telephone calls, multimedia, etc. Thus, the service requires data transmission, such as a telephone call or the transmission of multimedia streams, between the wireless communication device and the stationary parts of the system. One important task of the operation of a third-generation mobile communication system is to control (initialize, maintain and terminate, according to the need) bearer services in such a way that each requested service can be allocated to mobile stations without wasting the available bandwidth.

[0006] The quality of service determines, for example, how protocol data units (PDU) are processed in the mobile communication network during the transmission. For example, QoS levels defined for connection addresses are used for controlling the transmission order, buffering (packet strings) and rejecting packets in support nodes and gateway support nodes, particularly when two or more connections have packets to be transmitted simultaneously. The different QoS levels determine, for example, different delays for packet transmissions between the different ends of the connection, as well as different bit rates. Also, the number of rejected and/or lost packet data units may vary in connections with different QoS levels.

[0007] It is possible to request for a different QoS for each PDP context. For example, in e-mail connections, a relatively long delay can be allowed in the transmission of streams. However, real-time interactive applications, such as video conferencing, require packet transmission at a high rate.

[0008] In some applications, such as file transfers, it is important that the packet switched transmission is flawless, wherein in error situations, the packet data units are retransmitted, if necessary.

[0009] For the packet switched communication service in the UMTS system, the defining of four different traffic classes has been proposed, and for the properties of these traffic classes, the aim has been to consider the different criteria for the different connection types. One criterion defined for the first and second classes is that the transmission takes place in real time, wherein the transmission must have no significant delays. However, in such classes, the

accuracy of the data transfer is not such an important property. In a corresponding manner, non-real time data transmission is sufficient for the third and fourth traffic classes, but a relatively accurate data transmission is required of them. An example of real-time first-class communication is the transmission of conversational speech signals in a situation in which two or more persons are discussing with each other by means of wireless communication devices. An example of a situation in which real-time second-class communication might be feasible, is the transmission of a video signal for immediate viewing (streaming). Third-class non-real time packet communication can be used, for example, for the use of database services, such as the browsing of Internet home pages, in which the relatively accurate data transmission at a reasonable rate is a more important factor than the real-time data transmission. In the system according to this example, for example the transfer of e-mail messages and files can be classified to the fourth category. Naturally, the number of traffic classes is not necessarily four as mentioned here, but the invention can be applied in packet switched communication systems comprising any number of traffic classes. The properties of the four presented traffic classes are briefly presented in Table 1.

[0010] The guaranteed bit rate is used for admission control and resource reservation at the RAN and CN, the maximum bit rate is used for policing at the CN, i.e. no higher than the maximum bit rate is allowed to enter the CN at the GGSN, packets that exceed this bit rate will be dropped.

[0011] Modern second and third generation wireless communication devices have much better data processing properties than older wireless communication devices. For example, they already have the facility of connecting to the Internet and using a browsing application in the wireless communication device to retrieve information from the Internet, and in the future, it will be possible to set up multimedia calls, for example, for real-time video conferences and the like.

[0012] The requirements of different applications may be significantly different. Some applications require fast communication between the sender and the receiver. These applications include, for example, video and telephone applications. Some other applications may require as accurate data transmission as possible, but the bit rate of the data transmission connection is less important. These applica-

TABLE 1

	Class			
	First class (conversational class): real-time, e.g. telephone conversation guaranteed capacity no acknowledgement	Second class (streaming class): real-time, e.g. video information guaranteed capacity acknowledgement possible buffering on application level	Third class (interactive class): interactive best effort method acknowledgement Internet browser, Telnet real-time control channel	Fourth class (background class): background transmission by the best effort method acknowledgement background loading of e-mail messages, calendar events, etc.
Maximum bit rate (kbps)	<2048	<2048	<2048 - overhead	<2048 - overhead
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum packet size (Bytes) (SDU)	≤ 1500 or 1502	≤ 1500 or 1502	≤ 1500 or 1502	≤ 1500 or 1502
Transmission of incorrect packets (SDU)	Yes/No/—	Yes/No/—	Yes/No/—	Yes/No/—
Residual bit error ratio	$5 * 10^{-2}$, 10^{-2} , $5 * 10^{-3}$, 10^{-3} , 10^{-4} , 10^{-5} , 10^{-6}	$5 * 10^{-2}$, 10^{-2} , $5 * 10^{-3}$, 10^{-3} , 10^{-4} , 10^{-5} , 10^{-6}	$4 * 10^{-3}$, 10^{-5} , $6 * 10^{-8}$	$4 * 10^{-3}$, 10^{-5} , $6 * 10^{-8}$
Packet error ratio (SDU)	10^{-2} , $7 * 10^{-3}$, 10^{-3} , 10^{-4} , 10^{-5}	10^{-1} , 10^{-2} , $7 * 10^{-3}$, 10^{-3} , 10^{-4} , 10^{-5}	10^{-3} , 10^{-4} , 10^{-6}	10^{-3} , 10^{-4} , 10^{-6}
Transmission delay (ms)	100 ms - maximum value	250 ms - maximum value		
Guaranteed bit rate (kbps)	<2048	<2048		
Traffic processing priority			1, 2, 3	
Allocation priority	1, 2, 3	1, 2, 3	1, 2, 3	1, 2, 3

tions include, for example, e-mail and database applications. On the other hand, these applications can be used in several wireless communication devices with different properties.

[0013] The user of the wireless communication device may be willing to watch a multimedia presentation with the wireless communication device. The user finds the loading address of such a presentation and sends a request to send the presentation to the wireless communication device. The request is handled in the communication system. The loading address of the requested multimedia presentation may address to a server in a communication network, such as a server of the Internet. The server which delivers the multimedia presentation to the receiving wireless communication device is called as a streaming server in this description.

[0014] The communication system should reserve enough resources for the communication between the streaming server and the wireless communication device to be able to deliver the requested multimedia presentation. Otherwise the presentation may not be presented with the same accuracy and error free in the receiving wireless communication device. In the UMTS communication system the wireless communication device requests a PDP context with certain QoS parameters first. Then, the network selects a bearer for the connection by using some selection bases, for example, the parameters the wireless communication device has possibly used in the request. Such selection bases may not be appropriate or accurate enough wherein situations may occur in which the bearer service can not provide enough transmission capacity for the connection, or it provides more capacity than is needed, wherein the usage of the network resources is not efficient.

[0015] Another situation in which a delivery of multimedia information may be needed is two wireless communication devices communicating with each other to exchange multimedia information such as video or still images. Also in this kind of situation enough resources should be reserved by the network for the communication. However, when using prior art methods it is not always possible to inform both ends of the connection about the demands for the connection.

[0016] Basic streaming systems are non-adaptive. For example, the current Packet Switched streaming Service (PSS) defined by 3GPP in releases 4 and 5 is non-adaptive. Packet Switched streaming Service in Rel. 6 will be adaptive. The adaptive characteristic is given by the ability of the system, i.e. both a streaming server and a client, to adapt to the varying network channel conditions such as changes in the QoS negotiated channel bit rates, transfer delays, other Quality of Service parameters, or even changes in the underlying network in case of handovers.

[0017] In order to make the system adaptive, some communication between the streaming server and clients must be established. This is already in place whenever the RTSP protocol is used for session set-up and control. However, the transmission of the necessary information between the server and the client must occur in a correct way in order to guarantee that the system is adaptive and ultimately the best user Quality of Service for audio and video streaming can be achieved.

[0018] For this purpose, some prior art techniques already enable the transmission of QoS information, coming from

the underlying mobile network, from a streaming client to a streaming server. This allows more cooperation between the two ends in order to make the system more adaptive.

[0019] What has not been specified so far is the relation between the QoS parameters in a specific mobile network environment and the PDP (Packet Data Protocol) context usage. For instance, different cases are possible. In the following, the associated RTCP flow related to each RTP media stream is not considered. Alternatively, considering the RTP and its associated RTCP flow as a part of the same multimedia stream does not change the nature of the problem:

[0020] 1. A PDP context carries only one media of a streaming session

[0021] 2. A PDP context carries all the medium of a streaming session in a case when there are more than one.

[0022] If the streaming client decides to signal to the streaming server e.g. via RTSP some of the QoS profile parameters, for example the guaranteed bit rate, the maximum bit rate or the transfer delay, some problems may occur to the server in the correct interpretation of the QoS profile and, in the end, in the nature of the network connection.

[0023] In RTSP there are two possible kinds of sessions, which are a so called aggregate controlled session and a non-aggregate controlled session. The aggregate controlled session is a session where, at the transport level, all media components can be controlled by a single command sent to the server by the client (e.g. one RTSP PLAY command for both audio and video components). If this does not happen, i.e. at least one media component is controlled individually in a session, then the session is said to have non-aggregate control.

[0024] In the following, some examples are disclosed to clarify the problems which relate to the negotiation of QoS parameters for multimedia streams. It should be noted that the examples and the different parameters used in the examples are non-restrictive and in practical implementations different kind of parameters and combinations of media streams may exist.

EXAMPLE 1

[0025] In this example the multimedia stream includes two medium (e.g. one audio stream and one video stream). All the different media are transmitted using a single PDP context.

[0026] It is supposed that the streaming client has received notification from the streaming server (e.g. via the SDP protocol), that the audio stream requires 12 kbps and the video bit stream requires 52 kbps. It is also supposed that the streaming client establishes a connection with the mobile network using a single PDP context, over which the client wishes to transmit both audio and video streams, and that the network has granted the PDP context with the following (among the others) QoS profile parameters:

[0027] Guaranteed bit rate=64 kbps

[0028] Maximum bit rate=70 kbps

[0029] Now, let's suppose that the streaming client wants to inform the streaming server about the granted QoS from

the network, in order to enable the system to be more adaptive. To increase the efficiency, the client is supposed to decide to signal this information before starting the playback of the two medium. Therefore, it chooses to signal the two fields above using the SETUP method. Since there are two media, the client will send the two fields embedded into two SETUP messages (one for audio and one for video), with the following information:

- [0030] SETUP (Audio):
- [0031] Guaranteed bit rate=12 kbps
- [0032] Maximum bit rate=70 kbps
- [0033] SETUP (Video):
- [0034] Guaranteed bit rate=52 kbps
- [0035] Maximum bit rate=70 kbps

[0036] The guaranteed bit rate signalled in each SETUP contains the required bandwidth for each media (which is known to both streaming server and streaming client), but the maximum bit rate information can only be the granted maximum bit rate in the PDP context. Therefore, it cannot be anything else than 70 kbps in this example, because there would be no way to split the maximum bit rate between the two media. The SETUP method is interpreted by the streaming server as being a per-media description. Therefore, the server will interpret as if there will be virtually two network channels with the characteristics described by the two SETUP messages (one channel with guaranteed bit rate of 12 kbps and a maximum bit rate of 70 kbps, and another channel with guaranteed bit rate of 52 kbps and a maximum bit rate of 70 kbps). The cumulative guaranteed bit rate of the media is $12+52=64$ kbps, which is the actual network guaranteed bit rate of the PDP context. The server is entitled to send a maximum bit rate of 70 kbps for audio and a maximum bit rate of 70 kbps for video. When a single PDP context is used, this means that the cumulative maximum bit rate of the medium is $70+70=140$ kbps, which is not the network maximum bit rate for the PDP context. Since each media stream can be transmitted at variable bit rate, the sum of the instantaneous bit rates of the two medium can reach 140 kbps in any moment of time. However, every value greater than maximum bit rate provided by the network (70 kbps in this example) is not allowed, because the network resources are not available. Therefore, the server is led to mis-interpret the QoS information of the PDP context. This drives to bad user QoS.

[0037] On the other hand, thinking of splitting the 70 kbps maximum bit rate in a proportional way between the two media, is something that would lead the server to make a sub-optimal usage of the channel usage, which is shared by the different medium. The server would try to use the channel as if there were two separate PDP contexts.

[0038] A similar problem occurs if the guaranteed bit rate information sent to the server is what is really granted by the network in the PDP context. For instance, if the information of the granted 64 kbps guaranteed bit rate is sent to the server in both SETUPS messages, even more problems would be generated because the server will be entitled to send at a guaranteed bit rate of even 64 kbps of audio and 64 kbps of video, making a total guaranteed bit rate of 128 kbps, which is not available in the PDP QoS in this example. This would produce network buffer overflow and bad user QoS.

EXAMPLE 2

[0039] In this other example the multimedia stream also includes two medium (e.g. one audio stream and one video stream), but every different medium is transmitted using separate PDP contexts.

[0040] It is supposed that the streaming client has received notification from the streaming server (e.g. via the SDP protocol), that the audio stream requires 12 kbps and the video stream requires 52 kbps. It is also supposed that the streaming client establishes a connection with the mobile network using two separate PDP contexts, over which the client wishes to transmit respectively the audio and video streams, and that the network has granted the PDP context with the following (among the others) QoS profile parameters:

- [0041] PDP context for Audio:
- [0042] Guaranteed bit rate=12 kbps
- [0043] Maximum bit rate=20 kbps
- [0044] PDP context for Video:
- [0045] Guaranteed bit rate=52 kbps
- [0046] Maximum bit rate=64 kbps

[0047] Now, let's suppose that the streaming client wants to inform the streaming server about the granted QoS from the network in order to enable the system to be more adaptive. The QoS information could be sent in a PLAY command. The PLAY command is generally interpreted by the server as being an aggregate session command. Therefore, only a couple of parameters must be sent. The client could decide to send a guaranteed bit rate= $12+52=64$ kbps, and a maximum bit rate= $20+64=84$ kbps. This confuses the server which will understand that a single PDP context is used with the specified QoS parameters, which is not the case in this example.

[0048] The streaming server is responsible for the delivery of streaming data and the wireless communication device prebuffers a pre-determined amount of data before the actual playback. The streaming server is also responsible for adjusting the transmission bit rate to compensate for the actual transmission bandwidth variation and maintaining the receiver buffer (sitter buffer).

[0049] In 3GPP the Packet Switch Streaming (PSS) clients can report the characteristics of the wireless communication network to the PSS server. The information can be included to the Real Time Streaming Protocol (RTSP) header, where parameters concerning the network are defined. The parameters can be e.g. network's guaranteed bit-rate, network's maximum bit-rate, and network's maximum transfer delay.

[0050] It is essential also for the multimedia streaming server to be able to adapt to different connection conditions and different network types, such as EGPRS (Enhanced General Packet Radio Service) and CDMA2000 (Code Division Multiple Access) 1xEV-DV. Different mobile networks behave differently, for example the EGPRS transmission channel bandwidth variation is smaller than that of a CDMA2000 1xEV-DV channel. If the streaming server knows about the type of the network the client is using, it can deliver the service better. For example if the client is on a high variation transmission channel, the streaming server

should not react too fast to the channel variation. On the other hand if the streaming server knows that mobile is operating on a stable channel then any channel variation should be taken into consideration and react faster. The streaming server can adjust the control parameters for the delivery of the streaming data if the network type is known. Unfortunately such information is only available on the client side.

[0051] Currently 3GPP and 3GPP2 have different naming conventions regarding the rate adaptation and other signalling. If the multimedia streaming server does not know what type of network the client is using, then during the signalling the server might have problem responding to the request. A 3GPP and 3GPP2 compliant server should be able to parse and send corresponding standard related signalling, but only if such information is available to the server.

[0052] In the above described examples cases the main problem is that the streaming server does not know what is the type of the network channel reserved for data transfer (that can be either single or multiple PDP context), because it has no visibility on the PDP context allocation type. This visibility is only at the streaming client side.

SUMMARY OF THE INVENTION

[0053] It is thus an aim of the present invention to present a method and a system for attempting to solve the possible misunderstandings that the server might encounter when informed by the client about the QoS information of the network PDP context(s).

[0054] Another aim of the current invention is to adjust the delivery of the streaming data and/or signalling to provide optimal quality of service.

[0055] Another aim of the current invention relates to a naming convention problem existing in the 3GPP and 3GPP2.

[0056] The aims of the invention are achieved by using different kind of parameter signalling methods for informing the server about the session properties granted to the client by the network.

[0057] The invention introduces signalling and a new header for wireless communication network type. The information of the wireless communication network type information is only available on the mobile. This invention propose to transmit this information from the receiver to the server.

[0058] The present invention has advantages when compared to systems and methods of prior art. The invention allows to make a streaming server aware of the QoS parameters granted for each PDP context. This allows a better and more accurate adaptation by specifying a more precise QoS profile parameters.

[0059] The invention allows also to make a streaming server aware of the network type used by wireless communication device, which further improves the quality of the service. Thanks to the invention the server may also properly construct 3GPP/3GPP2 compliant signalling.

[0060] The invention clears the conflict which occurs due to the single/multiple PDP context usage by the client for a streaming session, and the QoS Parameter signalling to the server.

[0061] If the procedure described in the invention is not used, then the multimedia session can not benefit from the QoS parameter information, but instead the quality of service risks to be severely degraded.

[0062] The invention makes use of the wireless streaming concepts and improves the multimedia streaming performance and adaptation for wireless domain, making use of the 3GPP and 3GPP2 specific protocols and codecs.

[0063] Another important advantage is given by the possibility to efficiently use the delta bandwidth computed as maximum bit rate—guaranteed bit rate. This bandwidth can be used for bandwidth adaptation or for handling peaks of video bit rate. Finally, this delta bandwidth can be used for delivering the best media quality when encoding multimedia streams in real-time, for example, by changing encoding parameters (including the media stream bit rate) on the fly that have impact on the bit rate.

DESCRIPTION OF THE DRAWINGS

[0064] In the following, the invention will be described in more detail with reference to the appended drawings, in which

[0065] FIG. 1 shows a system in which the method according to a preferred embodiment of the invention can be applied,

[0066] FIG. 2 shows a wireless communication device according to a preferred embodiment of the invention in a reduced block chart,

[0067] FIG. 3 shows a signalling diagram of QoS reservation and session control for a client with a single PDP context, and

[0068] FIG. 4 shows a signalling diagram of QoS reservation and session control for a client with multiple PDP context support.

DETAILED DESCRIPTION OF THE INVENTION

[0069] In the following description of a preferred embodiment of the invention, a UMTS type mobile communication system will be used as an example; however, it will be obvious for anyone skilled in the art that the invention is not limited solely to this system but it can also be applied in other communication systems (e.g. EGPRS) in which it is possible to determine various QoS levels for communication.

[0070] In the following the session description protocol (SDP) will be described in more detail.

[0071] On the Internet multicast backbone (Mbone), a session directory tool is used to advertise multimedia conferences and communicate the conference addresses and media-specific information necessary for participation. The multicast backbone is the part of the Internet that supports IP (Internet Protocol) multicast, and thus permits efficient many-to-many communication. It is used extensively for multimedia conferencing. Such conferences usually have the property that tight coordination of conference membership is not necessary; to receive a conference, a user at an multicast backbone site only has to know the conference's multicast group address and the UDP ports for the conference data streams.

[0072] Session directories assist the advertisement of conference sessions and communicate the relevant conference setup information to prospective participants. SDP is designed to convey such information to recipients. SDP is purely a format for session description—it does not incorporate a transport protocol, and can be conveyed with different protocols, including the Session Announcement Protocol, Session Initiation Protocol, Real-Time Streaming Protocol (RTSP), electronic mail using the MIME extensions, and the Hypertext Transport Protocol. SDP is intended to be general purpose so that it can be used for a wider range of network environments and applications than just multicast session directories.

[0073] A multimedia conference is a set of two or more communicating communication devices along with the software they are using to communicate. However, it will become evident that there can be other suitable applications, e.g. video conference.

[0074] A multimedia session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. A multimedia conference is an example of a multimedia session.

[0075] In the following some details of the present definitions of session description protocol will be described. Some descriptions of the protocol are required and some are optional. Optional items are marked with a '*'.

[0076] Session description

[0077] v=(protocol version)

[0078] o=(owner/creator and session identifier).

[0079] s=(session name)

[0080] i=* (session information)

[0081] u=* (URI of description)

[0082] e=* (email address)

[0083] p=* (phone number)

[0084] c=* (connection information—not required if included in all media)

[0085] b=* (bandwidth information)

[0086] One or more time descriptions (see below)

[0087] z=* (time zone adjustments)

[0088] k=* (encryption key)

[0089] a=* (zero or more session attribute lines)

[0090] Zero or more media descriptions (see below)

[0091] Time description

[0092] t=(time the session is active)

[0093] r=* (zero or more repeat times)

[0094] Media description

[0095] m=(media name and transport address)

[0096] i=* (media title)

[0097] c=* (connection information—optional if included at session-level)

[0098] b=* (bandwidth information)

[0099] k=* (encryption key)

[0100] a=* (zero or more media attribute lines)

[0101] According to the above mentioned document the bandwidth description is defined as follows:

[0102] b=<modifier>:<bandwidth-value>

[0103] This specifies the proposed bandwidth to be used by the session or media, and is optional.

[0104] <bandwidth-value> is in kilobits per second by default. Modifiers may specify that alternative units are to be used.

[0105] <modifier> is a single alphanumeric word giving the meaning of the bandwidth figure. Two modifiers are initially defined:

[0106] CT (Conference Total): If the bandwidth of a session or media in a session is different from the bandwidth implicit from the scope, a 'b=CT: . . . ' line should be supplied for the session giving the proposed upper limit to the bandwidth used. The primary purpose of this is to give an approximate idea as to whether two or more sessions can co-exist simultaneously.

[0107] AS (Application-Specific Maximum): The bandwidth is interpreted to be application-specific, i.e., will be the application's concept of maximum bandwidth. Normally this will coincide with what is set on the application's 'maximum bandwidth' control if applicable. For RTP based applications, AS gives the RTP "session bandwidth" as defined in section 6.2 of RFC 1889 (RTP) (including media bit rate and UDP/IP headers overhead).

[0108] The Real Time Streaming Protocol (RTSP) is a client-server protocol for controlling the delivery of data with real-time properties. It is used to establish and control either a single or several time-synchronized streams of continuous media, such as audio and video. RTSP is conveyed with transport protocols such as UDP and TCP. In other words, RTSP acts as a network remote control for multimedia servers. Sources of data can include both live data feeds (e.g. real-time video and/or audio) and stored clips (e.g. still images). An RTSP client and server negotiate an appropriate set of parameters for media delivery, partially using e.g. SDP (Session Description Protocol) syntax to describe those parameters.

[0109] FIG. 1 shows a part of the UMTS system, comprising a wireless communication device MT1, a radio access node 1 (RAN) which comprises a base station 2 (BS), and a radio network controller 3 (RNC) which controls the base station 2 and routes the connections between the base station 2 and the rest of the system, a wireless mobile switching centre 4 (WMSC) and a packet data access node 5 (PDAN) as routing possibilities in addition to the radio network controller 3. The UMTS system according to FIG. 1 also comprises e.g. a backbone network 6 and a packet data gateway 8 (PDG) to other packet networks, such as the Internet Protocol (IP) network 7, wherein the wireless communication device can communicate with e.g. a server 10 coupled to the IP network. Furthermore, FIG. 1 shows a circuit switched gateway 9 (Gateway to Mobile services Switching Centre, GWMSC) to couple to e.g. to a second

mobile communication network NW2, and a home location register 11 (HLR) e.g. to store the subscriber's access contract data.

[0110] Further, FIG. 2 shows, in a reduced block chart, a wireless communication device MT1 complying with a preferred embodiment of the invention, and which in this example is a communication device comprising data processing functions and mobile station functions, such as Nokia 9210i Communicator. The wireless communication device MT1 comprises e.g. one or more processors CPU, DSP, memory means MEM, the UMTS subscriber identity module (USIM) or corresponding means for identifying the subscriber, and a radio part RF for communication with the base station 2. The processor CPU can be integrated e.g. in an application specific integrated circuit 12 (ASIC), with which it is possible to perform a large number of the logical functions of the wireless communication device MT1. The memory means preferably comprise a random access memory (RAM), a read only memory (ROM), and at least part of the memory of the subscriber identity module USIM. The wireless communication device MT1 also comprises one or more user interfaces, preferably comprising a keypad 13, 14, a display device 15, 16, and audio means, e.g. a microphone 17, a speaker 18 and a codec 19.

[0111] In FIG. 1, it is assumed that the functions related to call management (CM) are implemented in the wireless communication device MT1 and in both the wireless mobile switching centre 4 and the packet data access node 5. These call management functions constitute the means for initializing, maintaining and terminating a call. Consequently, the wireless communication device MT1 and the wireless mobile switching centre 4 or the packet data access node 5 exchange call signalling messages to initialize, maintain and terminate a call. The functions of bearer management (BM) and radio resource management (RM) are implemented in the wireless communication device MT1 and in the radio network controller 3. The bearer management functions are utilized to select, for example, one or several logical channels according to the properties of the bearer service selected for communication between the wireless communication device MT1 and the base station 2, to provide a quality of service complying with the bearer service. The radio resource management functions are used, for example, to select the radio channel for the radio communication between the wireless communication device MT1 and the base station 2.

[0112] The packet data transmission connection between the wireless communication device MT1 and the IP network 7 can be set up from the packet data access node 5 (PDAN) via the packet data backbone 6 and the packet data gateway 8 (PDG). It is possible to set up a circuit switched data transmission connection between the wireless communication device MT1 and the mobile communication network via the radio access node 1, the wireless mobile switching centre 4 and the gateway to mobile services switching centre 9 (GWMSC). This gateway to mobile services switching centre 9 comprises means for setting up a connection between the mobile communication network and the second network NW2, such as GSM, PSTN or ISDN.

[0113] In the following, the method according to the a preferred embodiment of the present invention for streaming multimedia applications will be described with reference to

the system of FIG. 1 and the signalling diagrams of FIGS. 3 and 4. The following implementations are based on the usage of RTSP protocol. Also, the "QoSParams, MaxBW, GuaBW, TdelayMax and url" parameters are fictitious parameter names which are conceptual placeholders for the above-explained invention. They may be named differently in real-life implementations.

[0114] First, some terms will be defined. A Client is a wireless communication device MT1 and a Server is a streaming multimedia service provider (e.g. server 10 in FIG. 1) to the Client. A Multimedia Session is an interval of time during which multimedia related data is exchanged between the Client and the Server. A Multimedia Session Set-up Phase is the time interval during which the Client and the Server exchange multimedia session related set-up information, e.g. multimedia components to be used during the session, bandwidth information, multimedia codec related information, etc. A PDP context is the logical indication of an abstract bound between the QoS Resource reservation process and the mobile station running the streaming client.

[0115] The Client may be in a QoS (Quality of Service) enabled network NW1, which can provide some guarantees to the Client based on its resources. These guarantees may cover one or more of the following:

[0116] Maximum Bit rate (MaxBW): The Maximum bandwidth that can be used by the negotiated media component or the total multimedia session.

[0117] Guaranteed Bit rate (GuaBW): The bandwidth value that the QoS reservation procedure guarantees to the client for the negotiated media component or the total multimedia session.

[0118] Transfer delay (TDelayMax): the delay (in milliseconds) that each data unit experiences during the transmission from the server to the client and vice-versa.

[0119] Other parameters can also be defined but they are not described in detail here.

[0120] The invention covers the two different possibilities that the client can experience based on its ability to have multiple or single PDP contexts during a multimedia session.

[0121] First, the situation in which the client is only able to handle a single PDP context at a time will be described in more detail. In other words, a client with a single PDP context support can have a single QoS resource reservation at a single time, which spans all the media components (i.e. audio, video, etc) during the multimedia session. This means that the multimedia data, regardless of being a video or audio, etc. data, share the same transmission channel with the same QoS resources.

[0122] In the first scenario, an aggregate controlled session is activated for the wireless communication device MT1 (client) having only single PDP context support for the streaming session. In this scenario, if the wireless communication device MT1 has multiple media components to Set-Up for the session (e.g. an audio and also an accompanying video stream), then the client must not send the negotiated QoS parameters such as maximum bit rate MaxBW, guaranteed bit rate GuaBW, maximum transfer delay TdelayMax and any other QoS profile parameters to

the server **10** during the Set-Up phase because of the problem described in the background art section of this application.

[0123] The QoS negotiated parameters must be sent to the server at or after the transmission of the stream is activated i.e. at or after a Play command has been transmitted from the wireless communication device MT1 to the server **10**.

[0124] The command sequence could be as follows (FIG. 3):

[0125] The wireless communication device MT1 transmits **301** a Describe Session command to the server **10**.

[0126] DESCRIBE rtsp://server.com/session1.3gp
RTSP/1.0

[0127] CSeq: 1

[0128] Accept: application/sdp

[0129] The server **10** replies to this command by transmitting **302** SDP description including information on different media streams.

[0130] RTSP/1.0 200 OK

[0131] CSeq: 1

[0132] Content-Base: rtsp://server.com/session1.3gp/

[0133] Content-Type: application/sdp

[0134] Content-Length: 441

[0135] v=0

[0136] o=-3242987154 3242987154 IN IP4
111.111.111

[0137] s=session1.3gp

[0138] c=IN IP4 0.0.0.0

[0139] t=0 0

[0140] a=control:*

[0141] a=range:npt=0-60

[0142] m=video 0 RTP/AVP 96

[0143] b=AS:50

[0144] a=rtpmap:96 H263-2000/90000

[0145] a=control:trackID=2

[0146] a=range:npt=0-60

[0147] a=fmtp:96 profile=0;level=10

[0148] m=video 0 RTP/AVP 98

[0149] b=AS:40

[0150] a=rtpmap:98 H263-2000/90000

[0151] a=control:trackID=3

[0152] a=range:npt=0-60

[0153] a=fmtp:98 profile=0;level=10

[0154] m=audio 0 RTP/AVP 97

[0155] b=AS:10

[0156] a=rtpmap:97 AMR/8000/1

[0157] a=control:trackID=1

[0158] a=range:npt=0-60

[0159] a=fmtp:97 octet-align=1

[0160] m=audio 0 RTP/AVP 99

[0161] b=AS:20

[0162] a=rtpmap:99 AMR-WB/16000

[0163] a=control:trackID=4

[0164] a=range:npt=0-60

[0165] a=fmtp:99 octet-align=1

[0166] In the above SDP description, video1 has a b=AS definition of 50 kbps, video2 has a b=AS definition of 20 kbps, audio1 has a b=AS definition of 10 kbps and audio2 has a b=AS definition of 20 kbps.

[0167] Then, in the wireless communication device MT1 a selection is made, for example by the user, among the informed media to select the streams to be transmitted to the wireless communication device MT1. In this example it is assumed that video1 (50 kbps) and audio2 (20 kbps) are selected having a total bit rate of 70 kbps. After that the wireless communication device sends **303** a request for a bearer service to the communication network NW1. In the request the wireless communication device MT1 includes the wanted QoS parameters (maximum bit rate of 70 kbps) for all the media components.

[0168] In this example the network can only guarantee 60 kbps and enable a maximum bit rate of 80 kbps. Then, the network NT1 informs **304** the wireless communication device MT1 of the granted QoS parameters for the bearer service. After negotiating with the network for the bearer service for the PDP session, the wireless communication device MT1 transmits **305** a first setup message to the server **10** for informing the selected first medium stream, i.e. video1.

[0169] SETUP rtsp://server.com/session1.3gp/
trackID=2 RTSP/1.0

[0170] CSeq: 2

[0171] Transport:RTP/AVP/UDP;unicast;client-
_port=6984-6985;ssrc=31336d02

[0172] The server **10** replies **306** with OK message, if the selection is ok.

[0173] RTSP/1.0 200 OK

[0174] CSeq: 2

[0175] Session: 41

[0176] Transport:RTP/AVP/UDP;unicast;client-
_port=6984-6985;server_port=6900-6901;ssrc=
1d12115

[0177] The wireless communication device MT1 also transmits **307** a second setup message to the server **10** for informing the selected second medium stream, i.e. audio2.

[0178] SETUP rtsp://server.com/session1.3gp/
trackID=4 RTSP/1.0

[0179] CSeq: 3

[0180] Transport: RTP/AVP/UDP;unicast;client-
_port=6986-6987;ssrc=37115e8d

[0181] Session: 41

[0182] The server 10 replies 308 with OK message, if the selection is ok.

[0183] RTSP/1.0 200 OK

[0184] CSeq: 3

[0185] Session: 41

[0186] Transport: RTP/AVP/UDP;unicast;client_port=6986-6987;server_port=6902-6903;ssrc=7475313

[0187] The playback of the media streams is initiated by transmitting 309 a Play command from the wireless communication device MT1 to the server 10. In this case the Play command is included with information on at least the QoS parameters relating to the maximum bit rate and guaranteed bit rate which the network NT1 has granted.

[0188] PLAY rtsp://server.com/session1.3gp RTSP/1.0

[0189] CSeq: 4

[0190] Session: 41

[0191] QoSParams: MaxBW=80;GuaBW=60;TDelayMax=500

[0192] Range: npt=0-

[0193] The server replies to the command by sending OK to the wireless communication device MT1.

[0194] RTSP/1.0 200 OK

[0195] CSeq: 4

[0196] Session: 41

[0197] Range: npt=0-

[0198] RTP-Info:

[0199] url=rtsp://server.com/session1.3gp/trackID=2;seq=0;rtptime=10000,url=rtsp://server.com/session1.3gp/trackID=4;seq=0;rtptime=10000

[0200] Now, when the server 10 has received the Play command it knows that there exists a single QoS channel with QoS parameters signalled by the wireless communication device MT1 and the server 10 can adapt the transmission of the selected media streams according to the parameters.

[0201] After the Play command, the wireless communication device MT1 may update the negotiated QoS parameters for the whole multimedia session, using any other RTSP command defined within the context of the streaming system.

[0202] If the multimedia session is a non-aggregate controlled session (e.g. the audio and video data is retrieved from two separate servers), then the wireless communication device MT1 should not send the QoS parameters, as the separate media servers are not aware of each other, nor they are aware of the fact that the media components share the same QoS reserved channel.

[0203] Second, the situation in which the client is able to support multiple PDP contexts at a time will be described in more detail. In other words, a client with multiple PDP

context support can have multiple QoS resource reservations at a single time, which can be distributed between the media components (i.e. audio, video, etc) during the multimedia session. There can be a separate multimedia session for each media component (i.e. audio, video, etc) during the multimedia session. All the media components can have different QoS resource reservations.

[0204] In the second scenario, if the wireless communication device MT1 has multiple media components to setup for the session, and if the wireless communication device MT1 is willing to activate multiple PDP contexts for different media components, and also if the session control protocol does not allow media component url indicators to differentiate between media components, then the wireless communication device MT1 must not send the QoS negotiated MaxBW, GuaBW, TDelayMax and other QoS profile parameters to the server at the Play command, which will most probably be an addition of these parameters. The QoS parameters should instead be sent during the setup phase of each media component.

[0205] The command sequence could be as follows (FIG. 4):

[0206] The wireless communication device MT1 transmits 401 a Describe Session command to the server 10.

[0207] DESCRIBE rtsp://server.com/session1.3gp RTSP/1.0

[0208] CSeq: 1

[0209] Accept: application/sdp

[0210] The server 10 replies to this command by transmitting 402 SDP description including information on different media streams.

[0211] RTSP/1.0 200 OK

[0212] CSeq: 1

[0213] Content-Base: rtsp://server.com/session1.3gp/

[0214] Content-Type: application/sdp

[0215] Content-Length: 441

[0216] v=0

[0217] o=-3242987154 3242987154 IN IP4 111.111.111

[0218] s=session1.3gp

[0219] c=IN IP4 0.0.0.0

[0220] t=0 0

[0221] a=control:*

[0222] a=range: npt=0-60

[0223] m=video 0 RTP/AVP 96

[0224] b=AS:50

[0225] a=rtpmap:96 H263-2000/90000

[0226] a=control:trackID=2

[0227] a=range:npt=0-60

[0228] a=fmtp:96 profile=0;level=10

[0229] m=video 0 RTP/AVP 98

[0230] b=AS:40
 [0231] a=rtpmap:98 H263-2000/90000
 [0232] a=control:trackID=3
 [0233] a=range:npt=0-60
 [0234] a=fmtp:98 profile=0;level=10
 [0235] m=audio 0 RTP/AVP 97
 [0236] b=AS:10a=rtpmap:97 AMR/8000/1
 [0237] a=control:trackID=1
 [0238] a=range:npt=0-60
 [0239] a=fmtp:97 octet-align=1
 [0240] m=audio 0 RTP/AVP 99
 [0241] b=AS:20
 [0242] a=rtpmap:99 AMR-WB/16000
 [0243] a=control:trackID=4
 [0244] a=range:npt=0-60
 [0245] a=fmtp:99 octet-align=1

[0246] In the above SDP description, video1 has a b=AS definition of 50 kbps, video2 has a b=AS definition of 20 kbps, audio1 has a b=AS definition of 10 kbps and audio2 has a b=AS definition of 20 kbps.

[0247] Then, in the wireless communication device MT1 a selection is made, for example by the user, among the informed media to select the streams to be transmitted to the wireless communication device MT1. In this example it is assumed that video1 having a bit rate of 50 kbps and audio2 having a bit rate of 20 kbps are selected. After that the wireless communication device sends 403 a first request for a first bearer service to the communication network NW1. In the request the wireless communication device MT1 includes the wanted QoS parameters (guaranteed bit rate of 50 kbps) for the first media component (video1). In this example the network can only guarantee 50 kbps and enable a maximum bit rate of 80 kbps. Then, the network NT1 informs 404 the wireless communication device MT1 of the granted QoS parameters for the first bearer service. Next, the wireless communication device sends 405 a second request for a second bearer service to the communication network NW1. In the request the wireless communication device MT1 includes the wanted QoS parameters (guaranteed bit rate of 20 kbps) for the second media component (audio1). In this example the network can only guarantee 20 kbps and enable a maximum bit rate of 40 kbps. Then, the network NT1 informs 406 the wireless communication device MT1 of the granted QoS parameters for the second bearer service. After negotiating with the network for the bearer services for the PDP sessions, the wireless communication device MT1 transmits 407 a first setup message to the server 10 for informing the selected first medium stream, i.e. video1.

[0248] SETUP rtsp://server.com/session1.3gp/
 trackID=2 RTSP/1.0
 [0249] CSeq: 2
 [0250] Transport: RTP/AVP/UDP;unicast;client-
 _port=6984-6985;ssrc=31336d02

[0251] QoSParams:url=rtsp://server.com/
 session1.3gp/trackID=2;MaxBW=80;GuaBW=
 50;TDelayM

[0252] ax=500

[0253] The server 10 replies 408 with OK message, if the selection is ok.

[0254] RTSP/1.0 200 OK

[0255] CSeq: 2

[0256] Session: 41

[0257] Transport: RTP/AVP/UDP;unicast;client-
 _port=6984-6985;server-port=6900-6901;ssrc=
 1d12115

[0258] The wireless communication device MT1 also transmits 409 a second setup message to the server 10 for informing the selected second medium stream, i.e. audio2.

[0259] SETUP rtsp://server.com/session1.3gp/
 trackID=4 RTSP/1.0

[0260] CSeq: 3

[0261] Transport: RTP/AVP/UDP;unicast;client-
 _port=6986-6987;ssrc=37115e8d

[0262] Session: 41

[0263] QoSParams: MaxBW=40 ;GuaBW=20;TDe-
 layMax=500

[0264] The server 10 replies 410 with OK message, if the selection is ok.

[0265] RTSP/1.0 200 OK

[0266] CSeq: 3

[0267] Session: 41

[0268] Transport: RTP/AVP/UDP;unicast;client-
 _port=6986-6987;server-port=6902-6903;ssrc=
 7475313

[0269] The playback of the media streams is initiated by transmitting 411 a Play command from the wireless communication device MT1 to the server 10.

[0270] PLAY rtsp://server.com/session1.3gp RTSP/
 1.0

[0271] CSeq: 4

[0272] Session: 41

[0273] Range: npt=0-

[0274] In this case the Play command is not included with information on the QoS parameters relating to the maximum bit rate and guaranteed bit rate which the network NT1 has granted.

[0275] The server replies to the command by sending OK to the wireless communication device MT1.

[0276] RTSP/1.0 200 OK

[0277] CSeq: 4

[0278] Session: 41

[0279] Range: npt=0-

[0280] RTP-Info:

[0281] url=rtsp://server.com/session1.3gp/trackID=2;seq=0;rtptime=10000,url=rts

[0282] p://server.com/session1.3gp/trackID=4;seq=0;rtptime=10000

[0283] Alternatively, in the RTSP PLAY request, the wireless communication device MT1 could have done the following:

[0284] PLAY rtsp://server.com/session1.3gp RTSP/1.0

[0285] CSeq: 4

[0286] Session: 41

[0287] Range: npt=0-

[0288] QoSParams: url=

[0289] rtsp://server.com/session1.3gp/trackID=2;MaxBW=80;GuaBW=50;TDelayM

[0290] ax=500,url=

[0291] rtsp://server.com/session1.3gp/trackID=4;MaxBW=40;GuaBW=20;TDelayM

[0292] ax=500

[0293] Now, the server can identify which QoS parameters are assigned with which media component, based on its media component URL.

[0294] Server→Client: OK

[0295] RTSP/1.0 200 OK

[0296] CSeq: 4

[0297] Session: 41

[0298] Range: npt=0-

[0299] RTP-Info:

[0300] url=rtsp://server.com/session1.3gp/trackID=2;seq=0;rtptime=10000,url=rts

[0301] p://server.com/session1.3 gp/trackID=4;seq=0;rtptime=10000

[0302] Now, when the server 10 has received the Play command it knows that there exists multiple QoS channels with individual QoS parameters signalled by the wireless communication device MT1. As each media component will have its own set of QoS negotiated parameters valid for each PDP context, the server can safely associate each media component to the correct QoS negotiated channel, with the correct values assigned.

[0303] The client may also provide Information about the wireless communication network to the Server, which does not necessarily have this information because it is available on the Client side. According to the invention the information about network type is transmitted from the Client to the Server. For doing that, the invention provides a new RTSP header, "Mobile-Link-Char" and signalling for that.

[0304] As said, the "Mobile-Link-Char" header is defined for enabling the multimedia streaming service client to report the network characteristics to the streaming server. The "Mobile-Link-Char" header can be included in a request using any of the following methods: SETUP, PLAY, OPTIONS and SET_PARAMETER. The header can contain

one or more characteristics specifications. Each specification contains a URI that can either be an absolute or a relative, any relative URI use the RTSP request URI as base. The URI points out the media component that the given parameter apply to. This can either be an individual media stream or a session aggregate. The "Mobile-Link-Char" header should be included in a SETUP or PLAY request by the client, to give the initial values for the link characteristics. A SET_PARAMETER or OPITIONS request can be used to update the Mobile-Link-Char values in a session currently playing.

[0305] Into the "Mobile-Link-Char" a field for mobile network type, MNT, is included, which defines the wireless communication network used for signalling. Along with "MNT" there can be also other parameters such as GBW, for the forward link user data rate in kbits/s; and MTD, for the forward link maximum delay in milliseconds; and MBW, for the forward link maximum bit rate in kbit/s. Also some other parameters than the ones being mentioned can be included to the header.

[0306] The "MNT" field according to the invention is composed according to the following example:

[0307] MNT=<Standard body>-<Network type ID>-<Release Information>

[0308] The "standard body" is used for identifying the corresponding network, e.g. 3GPP or 3GPP2. This allows the server to construct suitable signalling, which is compliant the network in question.

[0309] The valid strings in the "MNT" network type field can be, e.g. EGPRS, W-CDMA, CDMA2000 and "MNT" release information are relevant release versions of network. For 3GPP "Network type ID" corresponds to EGPRS or W-CDMA (Wideband Code-Division Multiple-Access) and for 3GPP2 to HRPD (High Rate Packet Data) or SSS (Spread Spectrum Systems). For 3GPP "Release information" corresponds to REL-x (where x is a number) and for 3GPP2 to REL-y (where y is 0, A, B, C, etc.). Fields for "Standard body" and "Network type ID" are listed only for reference purpose. It is obvious that newer additions can be made into those without departing from the scope of the invention.

[0310] For example for the 3GPP2 compliant network (1xEV-DV) the header is:

[0311] Mobile-Link-Char: url="rtsp://server.example.com/media.3g2";

[0312] MNT=3GPP2-SSS-REL-C

[0313] Another example is a 3GPP2 compliant network (1xEV-DO), where the header is:

[0314] Mobile-Link-Char: url="rtsp://server.example.com/media.3g2";

[0315] MNT=3GPP2-HRPD-REL-0

[0316] Yet another example is a 3GPP compliant network, where the header is

[0317] Mobile-Link-Char: url="rtsp://server.example.com/media.3gp";

[0318] MNT=3GPP-EGPRS-REL-5

[0319] Yet another example is a 3GPP (1xEV-DV) compliant network, where the header with other parameters (bit rate is 32 kbps and maximum transfer delay is 2.0 seconds) is

[0320] Mobile-Link-Char: url="rstp://server.example.com/media.3gp";

[0321] MNT=3GPP-CDMA2000-Rev-C; GBW=32; MTD=2000

[0322] The Server can deduct from the "MNT" field which of the 3GPP or 3GPP2 compliant network the client is operating on. Due to this it can respond to the client with proper syntax for RTSP, DP, RTCP and other signalling. The terms 1xEV-DV and 1xEV-DO correspond to an evolution of a data transfer (Data and Voice, Data Only correspondingly).

[0323] If a QoS re-negotiation occurs for a particular PDP context (i.e. a particular media component), the client can signal the new QoS values using any of the available RTSP commands, by correctly referencing the media component for which the changes has occurred.

[0324] If the session control protocol allows media component url indicators to differentiate between media components, then the QoS Parameters can be signalled at the Play request too. The following pseudo-command sequence shows the possible scenario:

[0325] Client→Server: Setup (media component 1)

[0326] Server→Client: OK

[0327] Client→Server: Setup (media component 2)

[0328] Server→Client: OK

[0329] Client→Server : Play (URL of media component 1+ Negotiated QoS parameters for the media component 1; URL of media component 2+ Negotiated QoS parameters for the media component 2)

[0330] Server→Client: OK

[0331] In the above example, the server can differentiate between the media components and the QoS Parameters assigned for each component by the usage of the "media component URL" information. This field is a unique identifier of the media component in a session. If the client and the server can make use of such a parameter, then the wireless communication device MT1 may choose to send the QoS Parameters either at the Set-Up phase, or at the Play phase. This media component URL indicator also gives the possibility to the wireless communication device MT1 to update the QoS Parameters during the session, if a QoS re-negotiation occurs.

[0332] If the multimedia session is a non-aggregate controlled session (e.g. the audio and video data is retrieved from two separate servers), then the client can safely signal the QoS negotiated parameters at the Set-Up command, as well as the Play command, since there will be separate Play commands for each media component.

[0333] The media component URL field may also be present to identify the session URL in the first example but the restriction on not sending the QoS Parameters at Set-Up phase is still valid for that case.

[0334] If a QoS parameter set does not contain the media component URL, then the request URL of the streaming control protocol must be used as the main URL for QoS parameter assigning.

[0335] In the description RTSP is used as preferred method. It will become evident that other external methods are also possible. The example described here only provided a sample of part of the signalling, but other forms of such signalling are still in the scope of this invention. For understanding this, it will become evident, that also parameter names as well as network types can vary depending on the situation.

[0336] It is obvious that the present invention is not limited solely to the above-presented embodiments but it can be modified within the scope of the appended claims.

What is claimed is:

1. A method in a communication system comprising:

selecting at least one media stream to be transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network;

defining QoS requirements for transmitting said selected at least one media stream;

reserving transmission resources from the wireless communication network for the transmission of said at least one media stream;

performing a setup procedure between the receiving communication device and the sending communication device for activating one packet data transmission connection;

requesting by the receiving communication device the start of transmission of the at least one media stream; and

transmitting the selected at least one media stream from the sending communication device to the receiving communication device, the transmitting comprising:

using one data transmission context in the transmission of the selected at least one media stream, wherein information on the reserved resources is transmitted to the sending communication device at or after said requesting the start of transmission of the at least one media stream.

2. A method according to claim 1 comprising defining at least the following parameters for the data transmission connection:

maximum bit rate,

guaranteed bit rate,

transfer delay;

and informing said parameters to the sending communication device.

3. A method according to claim 1, comprising transmitting of information on the wireless communication network type to the sending communication device at or after the start of transmission of the at least one media stream requested by the receiving communication device.

4. A method in a communication system comprising:
 - selecting at least one media stream to be transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network,
 - defining QoS requirements for transmitting said selected at least one media stream,
 - reserving transmission resources from the wireless communication network for the transmission of said at least one media stream,
 - performing a setup procedure between the receiving communication device and the sending communication device for activating at least one packet data transmission connection,
 - requesting by the receiving communication device the start of transmission of the at least one media stream is requested, and
 - transmitting media streams from the sending communication device to a receiving communication device, the transmitting comprising
 - using one data transmission context for each selected media stream, wherein information on the reserved resources is transmitted to the sending communication device in connection with the setup procedure.
5. A method in a communication system, in which media streams are transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network, wherein at least one media stream is selected to be transmitted to the receiving communication device, type of the wireless communication network for transmitting said selected at least one media stream is defined, transmission resources are reserved from the wireless communication network for the transmission of said at least one media stream, a setup procedure is performed between the receiving communication device and the sending communication device for activating one packet data transmission connection, and the start of transmission of the at least one media stream is requested by the receiving communication device, and one data transmission context is used in the transmission of the selected at least one media stream, wherein information on the wireless communication network type is transmitted to the sending communication device at or after the start of transmission of the at least one media stream requested by the receiving communication device.
6. A method according to claim 5, wherein at least a parameter for defining the wireless communication network type is defined for the data transmission connection.
7. A method according to claim 6, wherein at least information about network standard, network type and network release is defined.
8. A method in a communication system comprising:
 - requesting by a receiving communication device information on QoS requirements for transmitting at least one media stream from a sending communication device to the receiving communication device at least partly via a wireless communication network;
 - requesting by the receiving communication device transmission resources from the wireless communication network for the transmission of said at least one media stream;
 - reserving by the wireless communication network resources for the transmission and transmits information on the reserved resources to the receiving communication device;
 - performing by the receiving communication device and the sending communication device a setup procedure for activating the one packet data transmission connection;
 - requesting by the receiving communication device the start of transmission of the at least one media stream by transmitting a start of transmission command to the sending communication device in which command also information on the reserved resources are transmitted to the sending communication device; and transmitting media streams from the sending communication device to the receiving communication device using one packet data transmission connection.
9. A method in a communication system comprising:
 - requesting by a receiving communication device information on QoS requirements for transmitting at least one media stream from a sending communication device to the receiving communication device at least partly via a wireless communication network using at least one packet data transmission connection;
 - requesting by the receiving communication device transmission resources from the wireless communication network for the transmission of said at least one media stream;
 - reserving by the wireless communication network resources for the transmission;
 - transmitting information on the reserved resources to the receiving communication device;
 - performing by the receiving communication device and the sending communication device a setup procedure for activating the at least one packet data transmission connection;
 - the setup procedure comprising:
 - transmitting by the receiving communication device information on the reserved resources to the sending communication device, and
 - requesting by the receiving communication device the start of transmission of the at least one media stream by transmitting a start of transmission command to the sending communication device in which command no information on the reserved resources are transmitted to the sending communication device;
 - wherein the method further comprising transmitting media streams from the sending communication device to the receiving communication device.
10. A communication system comprising means for transmitting media streams from a sending communication device to a receiving communication device at least partly via a wireless communication network, the communication system further comprising:
 - means for selecting at least one media stream to be transmitted to the receiving communication device,
 - means for defining QoS requirements for transmitting said selected at least one media stream,

means for reserving transmission resources from the wireless communication network for the transmission of said at least one media stream,

means for performing a setup procedure between the receiving communication device and the sending communication device for activating one packet data transmission connection,

means for requesting the start of transmission of the at least one media stream by the receiving communication device,

means for using one data transmission context in the transmission of the selected at least one media stream, and

means for transmitting information on the reserved resources to the sending communication device at or after the start of transmission of the at least one media stream is requested by the receiving communication device.

11. A communication system according to claim 10 comprising means for transmitting information on the wireless communication network type to the sending communication device at or after the start of transmission of the at least one media stream requested by the receiving communication device.

12. A communication system comprising means for transmitting media streams from a sending communication device to a receiving communication device at least partly via a wireless communication network using at least one packet data transmission connection, the communication system further comprising:

means for selecting at least one media stream to be transmitted to the receiving communication device,

means for defining QoS requirements for transmitting said selected at least one media stream,

means for reserving transmission resources from the wireless communication network for the transmission of said at least one media stream,

means for performing a setup procedure between the receiving communication device and the sending communication device for activating the at least one packet data transmission connection,

means for requesting the start of transmission of the at least one media stream by the receiving communication device,

means for using one data transmission context in the transmission of the selected at least one media stream, and

means for transmitting information on the reserved resources to the sending communication device in connection with the setup procedure.

13. A communication system according to claim 12 comprising means for defining at least the following parameters for the data transmission connection:

maximum bit rate,

guaranteed bit rate,

transfer delay.

14. A communication system according to claim 12 comprising means for transmitting information on the wireless communication network type to the sending communication device at or after the start of transmission of the at least one media stream requested by the receiving communication device.

15. A sending communication device comprising at least:

means for receiving a request for reserving transmission resources from the wireless communication network for the transmission of at least one media stream;

means for reserving resources for the transmission of at least one media stream;

a transmitter for transmitting the at least one media stream to the receiving communication device, said transmitter being adapted to send a setup message in connection with a setup procedure between the receiving communication device and the sending communication device for activating the at least one packet data transmission connection;

means for using one data transmission context in the transmission of the selected at least one media stream; and

a receiver adapted to receive information on the reserved transmission resources from the wireless communication network in connection with the setup procedure.

16. A receiving communication device comprising at least:

a receiver for receiving media streams from a sending communication device transmitted at least partly via a wireless communication network using at least one packet data transmission connection;

means for selecting at least one media stream to be transmitted from the sending communication device to the receiving communication device;

means for defining QoS requirements for transmitting the selected at least one media stream;

means for requesting transmission resources from the wireless communication network for the transmission of said at least one media stream;

means for performing a setup procedure between the receiving communication device and the sending communication device for activating the at least one packet data transmission connection;

means for transmitting information on the reserved resources to the sending communication device in connection with the setup procedure;

means for requesting the start of transmission of the at least one media stream from the sending communication device; and

means for using one data transmission context in the transmission of the selected at least one media stream.

17. A wireless communication device comprising at least:

a receiver for receiving media streams from a sending communication device transmitted at least partly via a wireless communication network using at least one packet data transmission connection;

means for selecting at least one media stream to be transmitted from the sending communication device to the wireless communication device;

means for defining QoS requirements for transmitting the selected at least one media stream;

means for requesting transmission resources from the wireless communication network for the transmission of said at least one media stream;

means for performing a setup procedure between the wireless communication device and the sending communication device for activating the at least one packet data transmission connection;

means for transmitting information on the reserved resources to the sending communication device in connection with the setup procedure;

means for requesting the start of transmission of the at least one media stream from the sending communication device; and

means for using one data transmission context in the transmission of the selected at least one media stream.

18. A wireless communication device according to claim 17, wherein means for requesting transmission resources from the wireless communication network for the transmission of said at least one media stream comprises means for defining a setup message.

19. A wireless communication device according to claim 17, wherein means for requesting the start of transmission of the at least one media stream from the sending communication device comprises means for defining a play message.

20. A wireless communication device according to claim 17, comprising means for transmitting information on the wireless communication network to the sending communication device at or after the start of transmission of the at least one media stream requested.

21. A network element comprising at least:

means for receiving information on a selection of at least one media stream to be transmitted to a receiving communication device at least partly via a wireless communication network using at least one packet data transmission connection;

means for receiving QoS requirements for transmitting said selected at least one media stream;

means for requesting transmission resources from the wireless communication network for the transmission of said at least one media stream;

a transmitter for transmitting the selected at least one media stream to the receiving communication device, said transmitter being adapted to send a setup message in connection with a setup procedure between the receiving communication device and the network element for activating the at least one packet data transmission connection;

means for using one data transmission context in the transmission of the selected at least one media stream; and

a receiver adapted to receive information on the reserved transmission resources from the wireless communication network in connection with the setup procedure.

22. A method in a communication system, in which media streams are transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network, wherein at least one media stream is selected to be transmitted to the receiving communication device, QoS requirements for transmitting said selected at least one media stream are defined, transmission resources are reserved from the wireless communication network for the transmission of said at least one media stream, a setup procedure is performed between the receiving communication device and the sending communication device for activating one packet data transmission connection, and the start of transmission of the at least one media stream is requested by the receiving communication device, and one data transmission context is used in the transmission of the selected at least one media stream, wherein information on the reserved resources is transmitted to the sending communication device at or after the start of transmission of the at least one media stream is requested by the receiving communication device.

23. A method in a communication system, in which media streams are transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network, wherein at least one media stream is selected to be transmitted to the receiving communication device, QoS requirements for transmitting said selected at least one media stream are defined, transmission resources are reserved from the wireless communication network for the transmission of said at least one media stream, a setup procedure is performed between the receiving communication device and the sending communication device for activating at least one packet data transmission connection, and the start of transmission of the at least one media stream is requested by the receiving communication device, and one data transmission context is used for each selected media stream, wherein information on the reserved resources is transmitted to the sending communication device in connection with the setup procedure.

24. A method in a communication system, in which media streams are transmitted from a sending communication device to a receiving communication device at least partly via a wireless communication network using one packet data transmission connection, wherein the receiving communication device requests information on QoS requirements for transmitting at least one media stream from the sending communication device to the receiving communication device, the receiving communication device requests transmission resources from the wireless communication network for the transmission of said at least one media stream, wherein the wireless communication network reserves resources for the transmission and transmits information on the reserved resources to the receiving communication device, the receiving communication device and the sending communication device perform a setup procedure for activating the one packet data transmission connection, and wherein the receiving communication device requests the start of transmission of the at least one media stream by transmitting a start of transmission command to the sending communication device in which command also information on the reserved resources are transmitted to the sending communication device.

25. A method in a communication system, in which media streams are transmitted from a sending communication device to a receiving communication device at least partly

via a wireless communication network using at least one packet data transmission connection, wherein the receiving communication device requests information on QoS requirements for transmitting at least one media stream from the sending communication device to the receiving communication device, the receiving communication device requests transmission resources from the wireless communication network for the transmission of said at least one media stream, wherein the wireless communication network reserves resources for the transmission and transmits information on the reserved resources to the receiving communication device, the receiving communication device and the sending communication device perform a setup procedure

for activating the at least one packet data transmission connection, in which setup procedure the receiving communication device also transmit information on the reserved resources to the sending communication device, and wherein the receiving communication device requests the start of transmission of the at least one media stream by transmitting a start of transmission command to the sending communication device in which command no information on the reserved resources are transmitted to the sending communication device.

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