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Title: Requirements of IP based Multimedia-Applications over UMTS

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Introduction

The Internet and it's application the World Wide Web (WWW) is the most popular multimedia system of today. As a consequence it is broadly accepted that mobile access to Internet resources and its applications (video conferencing, video telephony, streaming audio and video, WWW, E-Mail, File Transfer, etc.) will bring a major push for the mobile data market. That has already been reflected in TS22.00 clearly stating that UMTS will interconnect to the IP-world.

Meeting QoS guarantees in such heterogeneous and distributed environment involving fixed and mobile networks and terminals is fundamentally an end – to – end issue, that is from application to application. IETF has already specified and tested mechanisms for interworking with other networks in order to guarantee end – to – end QoS, for UMTS shall adopt these mechanisms to establish interworking.

To guarantee end – to – end QoS within UMTS the three core areas: air – interface , access and core have to be aware of the requirements and support the end – to – end QoS context.

Classes of Applications

Traffic over the Internet significantly changed since the early days of the Net. In the beginning of the Internet the exchange of ASCII files was the only communication form. Today multimedia e-mail, Internet Telephony and extensively web pages are common. Also Internet radio and video multicast are in use today.

Due to the wide range of application requirements three different classes of applications were established:

- **Non-real time** – like e-mail. There are nearly no requirements in the network.
- **Streaming** – With streaming a nearly real time transport of voice, music and video is possible. Depending on the network load a appropriate play out buffer is filled to overcome congestion.
- **Real time** – high requirements on the network.

Real-time
For the real time class the requirements in delay and delay jitter are very high. More buffer lead to larger queuing delay. These delays are very disturbing even in interactive communication. Real-time is necessary for interactive communication like IP-Telephony and Videoconferencing. Real-time traffic is transmitted with UDP. Because of the real-time character a retransmission of lost packets is not useful, because packet retransmission would increase delay tremendous. Since real-time traffic is very sensitive against delay a Guaranteed Service will be the most desirable. A guaranteed delay and delivery is assured.

Streaming
Delay jitter is compensated by playout buffer with an impact on the delay. Some streaming applications also detect the network status and buffers an appropriate amount of data. Streaming is particularly used for audio and video transmission even for stored data as live feed. This applications are adapted very well to today Internet conditions but are limited because they can only react.
The streaming information is normally transported via UDP. Some vendors include a retransmission strategy in the application layer which make it possible to retransmit video key frames and inserts them into the data stream again.

Non Real-time
For non real-time applications there is no requirement in delay. Non real-time applications are the most used applications in the today used Internet (see also Traffic Patterns).
Non real-time traffic is transported with TCP. TCP is a reliable protocol which transmits each byte of information. This can take a longer time when retransmissions happen. In the today used IP networks these are the standard application and use only best effort. But in future a large number of scenarios are possible to favour important traffic to other (e.g. transmission of a X-ray image should be preferred to network games).

Example Applications
In the table below example applications are listed for the different types of application.

Class	Example application Bandwidth <64kbps	Bandwidth >64kbps	Bandwidth>2Mbps
Non-real-time	e-mail, http, news...	Urgent http	
Streaming	Streaming Audio	Streaming Video	Video on Demand, Video broadcast, high quality video
Real-time	IP-Telephony	Video, Picture Telephony, Videoconferencing	

Table 1: Example applications

Requirements of Applications to the Network
The requirements of multimedia applications to the network can be characterised by:

- bandwidth
- delay
- delay jitter
- bit error rate
- availability

These requirements have different weights for specific applications. As listed in table 2 the real-time applications have high requirements in delay and delay jitter. A streaming application can deal with delay jitter but is limited by the trade-off between delay and jitter and by the buffer size. Non real-time application can handle this but need for a good service appropriate bandwidth.
Availability is a often forgotten requirement and only remembered by mobile telephone users. Availability is very important for the normal communication services like telephony and e-mail.

Requirement	Application
Bandwidth	Important for all applications, especially for streaming and real-time applications
Delay	Video, Voice, IP Telephony (Real-time applications)
Delay jitter	Video, Voice, IP Telephony
Bit error rate	e-mail, fax, WWW, FTP,..(non real-time applications)
Availability	Important for all applications; especially for applications like IP Telephony, e-mail,...

Table 2: Special requirements for specific applications

For the different classes of applications, the requirements of real-time IP traffic are very high, especially delay and the appropriate bandwidth are absolute critical. With a too high delay interactive communication is nearly impossible.
For streaming delay and delay jitter in certain boundaries are not the problem because of buffering. But when the bandwidth is too low the best buffering can not generate an acceptable result.
For non real-time traffic high BER is a great problem. As non real-time applications use TCP as transport protocol a reliable, which means without loss, transmission is very difficult. With a high BER, likelihood of errors in a TCP packet rises and more retransmissions take place.

Traffic Patterns of single flows

As mentioned before audio and video transmissions are commonly used applications in today's internet although the quality is in most cases very poor. UMTS supports bandwidth up to 2 Mbps (in-house) and 384 (outdoor). This is not enough to transmit high quality video, but sufficient for videoconferencing.

Generally, there are two observable trends: on behalf of general improvements in codec quality, applications tend to use less and less bandwidth for a/v transmissions, while on the other home and business usage increases dramatically. Given that and the trend towards provision of IP QoS mechanisms, the real-time patterns can be expected to increase.
In table 3 some example applications are listed. This list is not complete and gives only a relation between the parameters.

	Bandwidth kbps	Delay	Delay jitter	Bit error rate	Comment
Video M-JPEG, Single picture compression, 300 x 300 pixels, 24 Bit colour	1000 700	Low	Low	10^{-4}	Picture with a high entropy, Picture with bureau surrounding
MPEG-1, low PAL-Quality	1500	Low	Low	10^{-4}	Bandwidth with free decoder up to factor 4, max. bit error rate 10^{-6}, that means in 18 minutes video min. 1 error.
MPEG-2, good TV-Quality	6000	Low	Low	10^{-4}	Online-coding problematic I, P, B Frames are added with different error protection
Picture telephone H.320	64	<200 ms	low	10^{-4}	Gateway functionality to the core network
Audio, Telephone	32	<200 ms	low	10^{-4}	Low Delay (Echo)
e-mail, FAX	10 - 64	N/A	N/A	10^{-6}	Delay independent
Internet, WWW	500 min 32	TCP time-out	TCP time-out	10^{-6}	Delay independent, for good usability smaller delay is required
Voice over IP	10 - 64	<200 ms			Needs low bandwidth but allows only small delay, delay jitter sensitive
Teleworking, Application-sharing,	2000	<100 ms		10^{-6}	Integration of different Components, Delay < 100 ms (ITU, G.114)

Table 3: QoS requirements of typical applications

Of course it is possible that one application combines e.g. video, voice and application sharing. Because of the fundamental difference between the requirements on QoS there will be different flows of traffic with different QoS parameters. To synchronise this flows protocols like RTP/RTCP can be used and are already tested and used in a number of applications.

Conclusion

Provision of QoS for multimedia applications is only meaningful with end – to – end negotiation, that is from application to application. This has to be considered for the complete UMTS system (Air-Interface, Access and Core) and interworking networks.

Network to network negotiation should be based on IETF mechanisms.

UMTS has to support three classes of applications: non-real time, streaming and real time.

Further requirements from an application perspective are:

- Usage of link layer features so UMTS, that guarantee
    - bit error rate
    - delay
    - delay jitter
    - availability
- Flexible Best Effort Service
- Adaptation of the TCP stack for the special requirements of wireless links
- Appropriate support for standard IP protocols like UDP, TCP etc.

It is important that these requirements are included in 22.05.