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# In House High Definition Multimedia: An Overview on Quality-of-Service Requirements

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### *Abstract*

The increasing demand for high definition multimedia applications in indoor environment requires a novel network design, which should be devised on a clear understanding of quality-of-service (QoS) requirements of the applications. In this paper we present an overview of the current and future High Definition (HD) multimedia applications, classify them into four categories with respect to their intrinsic QoS requirements. We discuss qualitatively and quantitatively all the QoS requirements for the specified application categories at the service and network levels respectively. We specifically consider in-house networks. We believe this work can help in providing guidelines for further design of the future home and office networks capable of HD multimedia provision fulfilling the users' requirements.[1]

## 1. Introduction

The advances in technologies of consumer electronics and an unprecedented growth in number of manufacturers of these electronic devices are leading a revolution in home entertainment and virtual office. In turn the communication needs of customers across the globe are also fueling this growth. After a decade of high growth in mobile personal devices the trend now is towards the future home and office environments where users are surrounded by high capacity devices capable of supporting high definition (HD) multimedia applications, which provide users a more satisfying audio-visual experience. Those devices can either be fixed installations of household devices such as LCD, Plasma HDTV displays, or the personal portable devices such as Laptops, video game consoles and the next generation DVD technologies such as HD-DVD or Blue-Ray. A commonly envisioned scenario in home is that "theatre-quality" videos and surrounding sound are brought to the user through a HDTV broadcasting system or from a multimedia content centre such as a Personal Video Recorder (PVR), providing high quality audio/video distribution throughout the house. We also expect emerging HD applications in the near future, such as, Telepresence [1] and 3-D virtual reality [2], which provide users an on-the-spot feeling despite the actual physical distance.

To enable HD multimedia in home and offices, new architectures, techniques and ideas have been proposed for building such networks on the emerging high capacity radio technologies, such as radio-over-fiber and 60 GHz radio [3]. Nevertheless, it is of great importance to have a clear view of the QoS requirements for the networks before setting hands on the detailed network design. However, the current literature on QoS requirements has not given much insights with respect to HD applications, which intrinsically have more stringent requirements on network performance compared to the prevailing VoIP or IPTV services. Thus in this paper we provide an overview and classification of different types of HD multimedia applications regarding their QoS requirements. Further, we quantify those requirements at the service and the network levels[2] in consensus with the earlier work on multimedia applications such as VoIP and IPTV.

The rest of the paper is organized as follows: Section 2 overviews the current and future HD multimedia applications. A classification of the applications is discussed in Section 3. Section 4 and Section 5 list qualitative and quantitative QoS requirements for the applications at the service and the network levels respectively. Section 6 concludes this article.

[1] This work was supported by Dutch IOP GenCom "Future Home Networks" project.

[2] We've proposed in [12] to use the term "level" to differentiate from the OSI model and to abstract two important functionalities such as service, which talks to the user; and the network, which facilitates communication supporting QoS by the network infrastructure.

## 2. Current and Future HD Multimedia Applications

We refer HD multimedia applications as the video, audio and still image applications with an increased display or visual resolution and higher quality stereo with multimodal sound. The examples of HD multimedia applications are given below.

- *HD Video*: refers to the video with a frame size greater than the standard definition video format which is 480 lines for NTSC and 576 lines for PAL/SECAM. Normally its resolution is 1080 or 720 lines and with a 16:9 aspect ratio (i.e., 1920 pixels×1080 lines or 1280 pixels×768 lines). Two types of scanning are used: the progressive scanning (p) or interlaced scanning (i). The number of frames per second can be 24, 25/50, 30/60. Two types of chroma subsampling schemes -- 4:2:2 with 10-bit quantization and 4:2:0 with 8-bit quantization are adopted, the latter aiming at saving the bandwidth. Table 1 gives the current types of HD Video and it also shows the data rate and storage needed for each type of HD video.
- *HDTV*: refers to the digital television broadcasting system in which the content is in HD video format. Due to differences of the power line frequencies used in different countries, the HD video format used in the US is 1280×720 pixels at 60 Hz, in Australia it is usually 1920/1440×1080 pixels at 50 Hz, and the format of the European HDTV satellite channel EURO1080 is 1920×1080 pixels, progressive scanning at the field rate of 50 Hz.
- *HD Gaming*: refers to the video game systems able to output an HD signal, up to 1080p, which are supported by the major video game providers. The recent advance in gaming technology also enables 3-D virtual reality in HD format [2].
- *HD Audio*: refers to the specifications released by Intel in 2004 for delivering HD audio [4], which is capable of playing back more channels at higher quality than previously integrated audio codecs . It supports up to eight channels at 192 kHz/32-bit audio quality, enables enhanced voice capture and provides improvements that support better jack retasking, for example, when a microphone is plugged into the speaker jack, the system can automatically re-route the microphone data to that jack.
- *HD Radio*: refers to a radio technology that enables AM and FM radio stations to broadcast their programs digitally in addition to providing their analog service of the past. It enables the FM stations with CD-quality sound and AM stations with FM-quality sound and transmission of digital information.
- *HD Photo*: refers to Microsoft's HD Photo file format, known as Windows Media Photo [5]. HD Photo combines both lossless and lossy image compression in the same design, and offers the increased image fidelity, preserves the entire original image content and enables high quality exposure and color adjustments of the image.

## 3. Classification of HD Multimedia Applications

The HD multimedia applications can be regarded as a set of evolved traditional (non HD) multimedia applications with the enhanced quality and the ease for further manipulating the content. Thus we retrospect the classification criteria used for the traditional multimedia applications and group HD multimedia applications into four categories as follows:

1) *HD Interactive*: applications that provide the real-time HD audio/video communications along with sharing applications in collaborative environment amongst multiple parties in a distributed environment, e.g. HD gaming, HD audio/video conferencing etc.
2) *HD Broadcasting*: applications that allow users to access the real-time delivery of HD multimedia content from a library of multimedia database by using a multimedia terminal, e.g. HDTV and HD radio.
3) *HD Streaming*: applications that provide the transfer of HD audio/video contents from a server to a client at the request of an end user, allowing the user viewing in real time and supporting additional commands such as fast forward, rewind, play-back and so on. The examples are HD audio/video on demand.
4) *HD Messaging*: applications that enable the end user communication, by exchanging the composed multimedia content, using "store and forward" method of transmission, e.g. HD photo message, HD audio/video downloading and viewing them at a later time.

## 4. Service Level QoS Requirements for HD Multimedia Applications

ISO defines QoS as "the ability of a network element to have some level of assurance that its traffic and service requirements can be satisfied". QoS can be considered either subjective or objective. On one hand, users can differ in their perception of what is good quality and what is not; on the other hand, QoS has to be parameterized and to be assigned with specific values in consensus for network operators to meet. Intrinsically, at the service level the parameters are more meaningful to be interpreted with the subjective requirements by the end-user. From the user's perspective the term, Quality of Experience (QoE) has been used instead of QoS to describe the perception of the users regarding how good the service is delivered. QoE is usually determined by several aspects such as equipment quality, human factors, environmental factors, and transport quality. Several testing methods have been proposed by ITU-T for subjective audio and video quality measurement as indicated in its recommendation P.800 and G.1070 [6].

Mapping the QoE of users into relevant service level QoS parameters allows service providers to handle and manage meaningful parameters for representation. Due to the inherent attributes of the individual application, different types of HD multimedia applications have different QoS parameters to fulfill QoE requirements. We present in Table 2 the service level QoS parameters for the four types of HD multimedia applications based on the suggestion in ITU-T recommendation G.1010 [7].

HD Interactive applications require very short delays for providing real-time interactions for the end users with the HD audio/video content. This delay is bounded by the reaction time (RT) of humans; mean auditory RT is in the range of 140-160 ms and the mean visual RT is in the range of 180-200 ms [8]. Due to the delay introduced by the codecs, for example, the coding delay of MPEG-2 contributing to half of the tolerable end-to-end delay [8], it is not practical to use the compressed HD audio/video content. In this case, the uncompressed application data rate is up to 1~2 Gbps, referring to Table 1. An additional requirement for applications like video conferencing is that the delay difference between audio and video must be within a certain limit to provide “lip-synchronization”. This also relies on the human perception capability, which is normally around 100 ms. Human eye can average out the picture to tolerate some loss of information, so that some degree of packet loss is acceptable depending on the specific video codec and the type of error protection scheme used. Assuming the use of packet loss concealment algorithms to minimize the effect of packet loss, the information loss is bound by 1%, as suggested by ITU G.1010.

HD Broadcasting applications can have less stringent delay requirement for displaying the audio/video content than HD interactive applications since no conversational element is involved. From the moment of requesting the content till it is delivered to the end user, delay of a couple of seconds should be tolerable to ensure the real-time nature of the content. But the delay introduced by system acknowledging the user’s request must be short enough to give the user the interactive experience that they are controlling, (e.g., fast forward, rewind, start, pause) the display in real-time. Since for HD content delivery the delay requirement is not so stringent that compression techniques can be applied to reduce the data rate, e.g., up to 80 Mbit/s for 1080i by MPEG-2. The effect of other parameters, such as packet loss and delay difference between audio and video are inherently dependent on human perception capability, and should be the same for the HD interactive applications too.

HD Streaming applications can have less stringent delay for displaying the audio/video content than HD broadcasting applications, since the HD content is not streamed in real-time. Therefore, up to tens of seconds of delay can be tolerable. Similar to HD broadcasting applications, the delay for system acknowledgement should be short enough to give the user the interactive experience.

HD Messaging applications are not considered as real time applications, thus the delay requirement is not necessarily guaranteed. The application required data rate varies according to the media type of the content of the message. In general up to Mbps should be adequate.

## 5. Network level QoS requirement for HD multimedia applications

Once the service level QoS parameters are set, they must be translated into those readable by protocols supporting QoS at the network level. Typically, throughput, delay, jitter, and packet loss are used as the representative QoS parameters to evaluate the network performance.

### *5.1 Throughput*

To support HD multimedia applications, the network layer throughput is required to be higher than the application data rate required at the service level, due to the protocol headers, e.g. 12Bytes for RTP, 20Bytes for IPv4 header or 40Bytes for IPv6 header, and the control overhead such as RREQ, RREP, etc., within the network level. The control overhead is dependent on the specific networking protocols. Thus required throughput at the network level should not be less than the sum of the required data rate at the service level and the protocol overheads.

### *5.2 Delay*

The overall delay in a HD multimedia communication in general comprises three parts: 1) the service level encoding/decoding and picture reordering delays for the HD media content compression; 2) interface delays for network adaptation between network and codecs, which is led by the encapsulation of the coded slices into transport entities of the network, e.g. RTP packets, and is suggested to be no more than 30ms; and 3) network delays further include the delay in queuing, processing, transmission and propagation of packets in the network. Queuing delay is due to packets being put in queues at the routers and waiting to be transmitted to the next hop node. The amount of delay that is introduced depends on the router’s queuing policy and the processor speed. Processing delay consists of the time used for operation such as looking up a route and changing the header. In case of multiple interfaces, when a packet arrives, the router may need to decide on which interface it should be sent out. This delay mainly depends on the complexity of the routing algorithms. Transmission delay is the amount of time taken by a node in the network to transport a packet on the media and is mainly determined by the medium access protocols. For fully

TABLE 1: Types of High Definition Video

| Type | Size | Frames per Second | Mbps | GB per hour |
|---|---|---|---|---|
| 720 24p | 1280×720 | 23.976 fps progressive | 354 (8-bit 4:2:0)<br>420 (10-bit 4:2:2) | 155<br>194 |
| 720 25p | 1280×720 | 25 fps progressive | 369 (8-bit 4:2:0)<br>461 (10-bit 4:2:2) | 162<br>203 |
| 720 25p | 1280×720 | 29.97 fps progressive | 442 (8-bit 4:2:0)<br>552 (10-bit 4:2:2) | 194<br>243 |
| 720 50p | 1280×720 | 50 fps progressive | 737 (8-bit 4:2:0)<br>922 (10-bit 4:2:2) | 324<br>405 |
| 720 60p | 1280×720 | 59.94 fps progressive | 884 (8-bit 4:2:0)<br>1105 (10-bit 4:2:2) | 388<br>486 |
| 1080 24p | 1920×1080 | 23.976 fps progressive | 795 (8-bit 4:2:0)<br>994 (10-bit 4:2:2) | 350<br>437 |
| 1080 25p | 1920×1080 | 25 fps progressive | 829 (8-bit 4:2:0)<br>1037 (10-bit 4:2:2) | 365<br>456 |
| 1080 30p | 1920×1080 | 29.97 fps progressive | 994 (8-bit 4:2:0)<br>1243 (10-bit 4:2:2) | 437<br>546 |
| 1080 50i | 1920×1080 | 50 fields per second/25 fps interlaced | 829 (8-bit 4:2:0)<br>1037 (10-bit 4:2:2) | 365<br>456 |
| 1080 60i | 1920×1080 | 60 fields per second/29.97 fps interlaced | 994 (8-bit 4:2:0)<br>1243 (10-bit 4:2:2) | 437<br>546 |

TABLE 2: Service level QoS parameters

| HD MM Category | Media Type | Typical data rates | Service level QoS | | | |
|---|---|---|---|---|---|---|
| | | | One-way delay | Delay variation | Information loss | Other |
| Interactive | HD Audio/video | 1 ~ 2 Gbps (uncompressed) | <150 ms (best)<br><400 ms (limit) | < 1 ms | < 1% PLR | A/V delay difference<br>-90 ms ~ +120 ms |
| Broadcasting | HD Audio/video | 1 ~ 2 Gbps (uncompressed)<br>< 100 Mbps (MPEG-2) | <200 ms (control)<br>< 2 s (content) | < 1 ms | < 1% PLR | A/V delay difference<br>-90 ms ~ +120 ms |
| Streaming | HD Audio/video | 1 ~ 2 Gbps (uncompressed)<br>< 100 Mbps (MPEG-2) | <200 ms (control)<br>< 10 s (content) | < 1 ms | < 1% PLR | A/V delay difference<br>-90 ms ~ +120 ms |
| Messaging | HD Audio/video/ Image | In the order of Mbps | | | < 1% PLR | |

distributed MAC protocols (e.g. IEEE802.11), the links are homogeneous in the sense that every node uses the same algorithm to determine transmission, thus the transmission delay has the same statistical properties for all the nodes. However, the piconet based MAC protocols (e.g., IEEE802.15.3 based 60 GHz MAC protocol [10]) are designed to guarantee QoS, thus it is more favorable for high data rate multimedia communications. In these cases the different roles played by the controller and devices result in varied chances for the nodes to access medium, and thus it should have
different abstraction for different links. Propagation delay is the time the packet propagates on the air; and due to the speed of light, this delay is normally ignored for the local wireless networks. Thus the network delay can be taken as the sum of queuing delay at each router, the transmission delay at each hop and the processing delay with regard to the routing protocol. The network delays, together with the coding delay and the interface delay should be controlled within the delay requirements at the service level as specified in Table 2.

To analytically model the delays at the network level for HD multimedia applications, the abstraction of the service level delay is achieved by modeling the RTP packet arrivals with certain distribution of the coding delay and interface delay. The queuing delay is generally bounded by the router buffer size, the mean and variation of the inter-arrival time and the size of the incoming RTP packets. The transmission delay can be estimated by the information captured from the MAC layer performance. With the knowledge of the queuing delay and the transmission delay, we can thus generate a vision for the proper routing solution for the system to meet the service level QoS delay requirement, by making use of the analytical model for routing protocols.

### *5.3 Jitter*

As the variation in the network delay, Jitter can be alleviated by implementing jitter buffer at the destination, usually dependent on the application. At network level the jitter across the network should be limited to a range which is manageable by the service level techniques to further cancel it. To quantify the jitter requirement at network level, we follow the suggestions in ITU-T Y.1541 [11] and take 50 ms for HD multimedia applications.

### *5.4 Packet Loss*

Packets can be lost while in transit in the network due to various reasons such as the packet collision on the air or router buffer overflow. Besides if the packets fail to arrive within a reasonable period of time at the receiver then it can be considered as lost.

TABLE 3: The Network Level QoS requirement for HD Multimedia Applications

| HD MM Category | Media Type | Throughput | IPTD | IPDV | IPLR | IPER | QoS class* |
|---|---|---|---|---|---|---|---|
| Interactive | HD Audio/video | > 1 ~ 2 Gbps (uncompressed) | <100 ms (best)<br><400 ms (limit) | 50 ms | $10^{-6}$ | $10^{-5}$ | 6 (best)<br>7 (limit) |
| Broadcasting | HD Audio/video | > 1 ~ 2 Gbps (uncompressed)<br>>= 100 Mbps (MPEG-2) | <100 ms (control)<br>< 2 s (MM content) | 50 ms | $10^{-6}$ | $10^{-5}$ | 6 (control)<br>- (MM content) |
| Streaming | HD Audio/video | > 1 ~ 2 Gbps (uncompressed)<br>>= 100 Mbps (MPEG-2) | <100 ms (control)<br>< 10 s (MM content) | 50 ms | $10^{-6}$ | $10^{-5}$ | 6 (control)<br>- (MM content) |
| Messaging | HD Audio/video /Image | In the order of Mbps | | | | | |

* Refer to ITU-T recommendation Y.1541

The judgment for threshold for whether to consider a late packet as a lost packet is basically dependent on the content that will be given to the audio/video devices. In general the threshold should be restricted by the sum of delay and jitter requirements. If the packet arrives, but is corrupted, it is counted as corrupted. With the help of error correction techniques, one can retrieve the content of the packet to certain level, depending on the location of the bits in error and whether the error is affecting the understanding of the useful information carried by the packet. Only those packets that can not be corrected after FEC are counted as packet loss.

The packet loss can happen for both the control packets and the application content packets. The loss of the control packets will result in more degradation of the session. Therefore, the requirement for the packet loss at network level should be much stricter than the service level requirement. For high bit rate applications, ITU-T Y.1541 network performance objectives set bounds for IP packet loss ratio (IPLR) to $10^{-6}$ and IP packet error ratio (IPER) to $10^{-5}$.

### *5.5 Summary for the network level QoS requirement*

In summary, we list the QoS requirement in Table 3, which specifies the bounds for the QoS parameters -- network throughput, IP packet transfer delay (IPTD), IP packet delay variation (IPDV), IP packet loss ratio (IPLR), and IP packet error ratio (IPER), for the four HD multimedia application categories. In the meantime, we also suggest a mapping of the requirement of these four HD multimedia application categories onto the seven QoS classes defined by ITU-T Recommendation Y.1541. We believe the parameters and values listed in Table 3 can serve as a starting point for network protocol design and evaluation, as we see the growing importance and favor for service oriented architecture.

## 6. Conclusions

It is evident that the world is moving towards high quality media transportation on the network. In this article we presented an overview of the current and future HD multimedia applications for indoor network applications. While classifying those applications into four categories with respect to their intrinsic QoS requirements, we discuss them qualitatively and quantitatively for the specified application categories respectively at service and network levels. In general, HD multimedia applications have the comparable delay and jitter requirements to — whereas more stringent requirements on throughput and packet loss than — the traditional multimedia applications such as VoIP and IPTV. We believe this work brings insights and helps in providing guidelines for further design of home and office networks on wireless infrastructure capable of delivering various HD multimedia services with satisfied quality.

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