HOW TO EVALUATE THE QUALITY OF SERVICE OF SATELLITE BASED CONTENT DELIVERY NETWORKS

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ABSTRACT

A content delivery network (CDN) circumvents the congested Internet by pushing important content to large caches at the network edges. Content, produced at single points of origin, is sent to multiple points at the network edges, constituting point-to-multipoint (multicast) connections. For international wide area networks, such a scheme is most effectively implemented by using the intrinsic multicasting capabilities of a satellite. In this paper we describe the European project CODIS, which aims at implementing and testing a satellite based content delivery network. We also describe a framework for measuring the quality of service of CODIS and define rules for digital television broadcasters, yielding optimal bitrates to use in order to guarantee customer satisfaction.

KEY WORDS

Computer Networks, Content Delivery Network, Satellite

1. INTRODUCTION

In the past, content delivery networks (CNDs) have become increasingly popular, as they are able to bypass the congested Internet. Multimedia streams flowing through the Internet often face at least one congested link, causing presentation stalls or decreased video quality for the audience. CDNs are based on a network infrastructure of their own, consisting of terrestrial links or satellite based communication. By reducing the number of hops between the content and its consumers, multimedia presentations are more likely to remain unaffected by cross traffic. Due to the complexity of the used applications and their necessary protocols, it is not easy to effectively measure the quality of service (QoS) of a CDN offered to its customers. However, it is this very benefit of an increased QoS that make customers pay for CDN services in the first place.

In this paper we describe the EU project CODIS, which aims at setting up and running a satellite based CDN. We further describe in detail the CODIS methodology for evaluating the QoS of such a CDN. The methodology consists of the metrics to be measured, the scenarios in which to measure and the measurement hard- and software. As preliminary results, we also define rules for broadcasters for finding optimal bitrates for digital television.

2. THE CODIS CDN

CODIS[1] is an IST project supported by the European Commission. The CODIS consortium, consisting of Alcatel Space, the French space agency CNES, the broadcasting research institutions Télédiffusion de France (TDF) and Institut für Rundfunktechnik (IRT), the measurement equipment manufacturer Rohde & Schwarz, the content management system provider Activia, and the Institute for Computer Science and Business Informatics of the University of Vienna, will setup, run and test a satellite based CDN using the novel satellite STENTOR constructed by Alcatel Space, containing an on-board DVB-S multiplexer. Uplinks to STENTOR are limited to approx. 9 Mbit/s, the downlink is limited to 38 Mbit/s.

The goal of the CODIS trials is to demonstrate the usefulness and QoS of such a CDN, which may be used, for example, by public broadcasters to bring their pre-recorded content near the end users, or which may directly send their content live from anywhere using STENTOR as a CDN entry point.

Fig. 1 shows the structure of the CODIS CDN. Satellite stations will be placed at Toulouse, Metz, Munich and Vienna, each being able to send and receive data to and from STENTOR, with the exception of Vienna, which will run in receive mode only. At each site, a remotely operated cache stores content at the network edges. The main applications using the CODIS CDN will follow the MPEG-2 based digital video broadcast (DVB) standard [24] and the emerging multimedia standard MPEG-4 [22] over IP. Terminals being able to show both DVB and MPEG-4 streams will be enhancements of the multimedia home platform (MHP)[2] standard. For CODIS we will use the Siemens Fujitsu Activity Box. Further applications

[1] http//www.codis-satellite.com/
[2] http://www.mhp.org/

running over CODIS include Web, FTP, and a CDN management software.

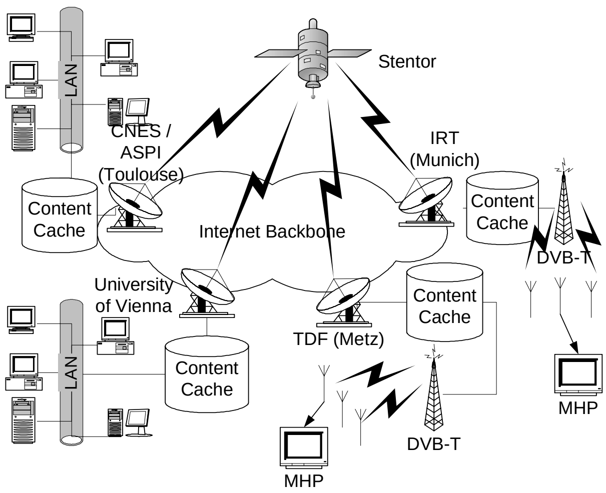

Fig. 1. The CODIS CDN network.

3. THE CODIS QOS FRAMEWORK

At a high level, our measurement framework is based on the CCITT Recommendation X.140 [12, 25], comprising a general framework for user-oriented QoS measures in data networks. In the following, we denote a measured statistic with *metric*.

For CODIS, we have defined QoS domains, representing different views of the network possible users may have. In the following, the QoS domains and the metrics to be measured for accessing and transporting content are explained within their QoS domain context.

3.1. IP Network Analysis

One major field of application for a CDN is to move data via TCP/IP. For measuring IP-related performance metrics, we start at the measurement framework [21] created by the IETF IP Performance Metrics (IPPM)[3] group. Let N be the number of links $l_1,\ldots,l_N$ interconnecting the nodes $h_0,\ldots,h_N$, b_i be the bandwidth of l_i, d_i be the length of l_i, c_i be the speed of signal transmission in l_i, f_i be the fixed amount of time used for incoming packets in h_i, and finally q_i be the non-deterministic queuing delay in h_i. Then the time it takes to send a message of size $s \le MTU$ (MTU denotes the maximum transfer unit, the largest packet a network can handle) over this path is given by

$$T_N(s) = \sum_{i=1}^{N} \frac{s}{b_i} + \frac{d_i}{c_i} + f_i + q_i \ . \qquad (1)$$

[3] http://www.ietf.org/html.charters/ippm-charter.html

Also, the *network latency*, the minimum time it takes to send (empty) messages over such a path is given by

$$L_N = \sum_{i=1}^{N} \frac{d_i}{c_i} + f_i + q_i \,, \qquad (2)$$

the time for sending an empty message to the receiver and back is called *round trip time* (*RTT*). Furthermore, the *bottleneck bandwidth* is given by

$$\hat{b} = \min\{b_1,\ldots,b_N\} \,. \qquad (3)$$

As CDNs move around large amounts of data, for $s > MTU$, instead of (1), we define the bulk transfer capacity [17] by

$$BTC_N(s) = s / T_N(s) \approx s / T_N^k(MTU) \,, \qquad (4)$$

as in this case the massage is split into k packets of size *MTU* (maybe with the exception of the last packet) and $T_N^k(MTU)$ denotes the time to move k packets of size *MTU* over the path. In this case, neglecting run-time situations by setting $q_i = 0$ and using (1), (2), and (3), the following bounds can easily be derived

$$L_N + \frac{kMTU}{\hat{b}} \le T_N^k(MTU) \le L_N + (k+N+1)\frac{MTU}{\hat{b}} \,. \quad (5)$$

(5) shows that the sending time can be bounded by just using network latency and bottleneck bandwidth.

Another limit is given for reliable transport protocols like TCP, which limit their sending rate by specifying the maximum number of bytes that may be sent without acknowledgement. For TCP this is set to 65535 bytes. This implies that for TCP the *stationary end-to-end bandwidth* is limited by $\bar{b} = 65535 / RTT$ bytes/s, and thus for *large* files

$$T_N(s) \ge L_N + s / \bar{b} \ . \qquad (6)$$

A summary of the IP metrics to be measured can be seen in Table 1.

Table 1: Codis IP network metrics.

Parameter	Symbol	Priority
Hop count	N	Mandatory
Network latency	L_N	Mandatory
Round trip time	RTT	Mandatory
Bottleneck bandwidth	$\hat{b}$	Mandatory
Max. TCP end-to-end bandwidth	$\bar{b}$	Mandatory
Packet loss rate	r_N	Mandatory
Bulk transfer capacity	BTC	Mandatory
Link latencies	$d_i / c + f_i$	Optional
Link bandwidths	b_i	Optional

A variety of tools exists which are able to measure the above described metrics.[4] We will mainly use iperf, Pchar, Chariot from NetIQ and self-written software.

[4] http://www.caida.org/tools/taxonomy/perftaxonomy.xml

3.2. CDN Performance Metrics

The CODIS CDN is based on IP. Consequently, the low-level ability of CODIS to transport content over IP is measured according to Table 1. On the other hand, CDNs have additional specific performance metrics that are to be addressed by CODIS. Publication time T_{pub} denotes the time necessary for pushing content from one cache to a number of others over the network. The cache hit rate C_h denotes the percentage of content that was served from the local cache instead of having been fetched from a remote location. The bandwidth out of cache C_{bw} means the bandwidth available from cache to end user. The CPU usage C_{cpu} means the relation of used CPU time versus available (total) CPU time. Finally, memory usage C_{mem} denotes the percentage of used memory versus total memory.

Table 2: CODIS CDN metrics.

Metric	Symbol
Publication time	T_{pub}
Cache hit rate	C_h
Bandwidth out of cache	C_{bw}
CPU usage	C_{cpu}
Memory usage	C_{mem}

Table 2 summarizes the CODIS CDN metrics to be measured. The single source for these metrics is the Activia content management software. Currently, the metrics are available via HTTP as HTML web pages or via the CDN log files.

3.3. MPEG-TS Analysis

One important aspect of CODIS is the transportation of digital television broadcasts. According to the DVB standard, the DVB data carrying multiple simultaneous programmes is transported in a so-called MPEG-2 transport stream (MPEG-TS). This multiplex contains programme information as well as the audio and video data of the broadcasts. In [9, 24] several metrics for measuring the consistency and quality of an MPEG-TS are described. The tests are grouped in three priorities according to the importance for monitoring purposes. The first group comprises a basic set of parameters that are necessary to ensure the transport stream can be decoded. Once first priority errors are detected it makes no sense to scan for second or third priority errors.
The second group comprises additional parameters which are recommended for continuous monitoring. Many of the tests in this group are only meaningful, if the content is not scrambled.
The third group lists optional additional parameters which could be of interest for certain applications. Most of the tests in this group refer to further tables with service information.
[9] proposes combinations of parameters described above, which can approximate the probability for a certain percentage of time and location that a service is available in a certain area with a defined quality. The Service_Availability_Error, Service_Degradation_Error and Service_Impairments_Error all work the same – they are the maximum of the number of occurrences of selected errors within a given time interval dT. Table 3 lists the types of errors used. As measurement equipment, we will use the Rohde & Schwarz tool Digital Video Measurement Decoder (DVMD), which can automatically decode and analyze MPEG-TS streams.

Table 3: Metrics for estimating availability of service.

Metric	Error Types
Service_Availability_Error	TS_sync_loss, PAT_error, PMT_error
Service_Degradation_Error	CRC_error, PCR_error, NIT_error, SDT_error
Service_Impairments_Error	Continuity_count_error, Transport_error

3.4. Response Time

In this methodology we use the term *access* to denote all direct interactions between users and the system. The main quality of service parameter is the user perceived *response time* T_r. Fig. 2 shows the definition of response time, being the time between issuing a request to the system until the result is visible (or audible) to the user. From intuition it is clear that longer response times decrease user satisfaction. It is, however, generally not easy to quantify the user satisfaction as a function of response time. In order to be able to interpret the response times as being acceptable or not, several limits have been proposed in the literature, which will be discussed briefly.

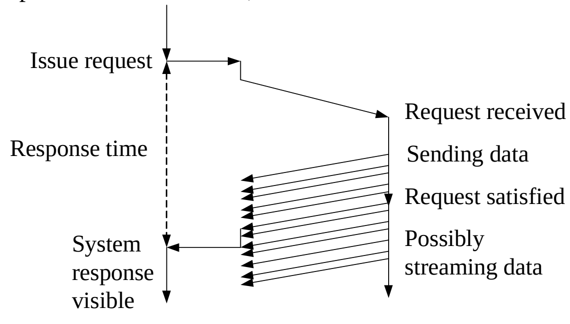

Fig. 2. Definition of response time.

Response time limits found in the scientific literature, denoting what users subjectively would rate “good”, include 4 seconds [23], 5 seconds [4], 8 seconds [32], 10 seconds [18], and 11 seconds [6]. In [5], Web response times have been rated for different scenarios using a scale *low*, *medium*, and *high*, describing also a general subjective rating of 30 individuals (Fig. 3). Two other metrics for direct interaction will be considered for CODIS. *Access denial probability* P_{ad} and *disengagement time* T_d, which will be interpreted like response time.

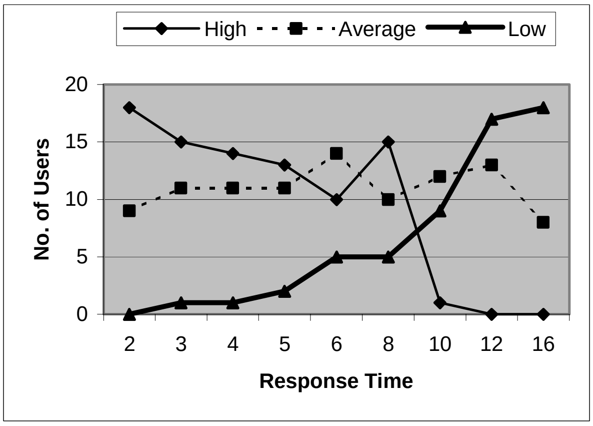

Fig. 3. Subjective rating of response time.

The above described parameters will be measured using the commercial tool Chariot, self-written Web and FTP robots initiating downloads automatically and a special version of the Siemens Fujitsu Activity Box.

Table 4: Codis user terminal access metrics.

Parameter	Symbol	Priority
Response time	T_r	Mandatory
Access denial probability	P_{ad}	Mandatory
Disengagement time	T_d	Optional

4 VIDEO TRANSPORT

If video data is sent over an unreliable data network, pictures might get distorted due to packet losses or dropped completely, or their presentation time might vary due to varying network traffic. An important property of a data network thus is given by its *temporal performance*, denoting the ability of a video transmission system to accurately reproduce motion or changing scenes [1]. For CODIS we have selected a number of basic transport metrics that we want to measure, here mainly for IP based MPEG-4 streaming [3].

The *frame rate* R_{fr} denotes the frames per second (fps) that are shown at the receiver. The interpretation of this metric will be based on the following facts: Standard cinema movies use 24 fps, PAL TV norm uses 25, and NTSC slightly less than 30 fps. Japanese cartoons use 18 fps, which is obviously regarded as absolute minimum for broadcasting. From cognitive science it is known that the human eye can not resolve frame rates larger or equal to 16 fps. According to [29], the minimum user requirement for frame rates is given by *5 frames per second* (absolute minimum).

Important transport metrics are given by the *overall bitrate* B_o and the *netto bitrate* B_n. The overall bitrate denotes the number of bits per second that are sent from a streaming server to the receiver. This includes also the lower protocol layers like RTP, UDP, TCP, and IP. The netto bitrate is the overall bitrate minus the bitrate used up by lower protocol layers. Whereas it is quite easy to measure the overall bitrate, for example by using a packet sniffer like windump, tcpdump, or ethereal,[5] measuring the netto bitrate requires an interpretation of the streams by the player, and thus requires to have access to player statistics.

Another transport metric is given by the *probability of a packet loss* P_{pl}. When using unreliable protocols like UDP, a lost packet may either cause audio or video impairments or must be resent, in case it transports important real-time data.

The probability that the presentation suddenly *stalls* is denoted by P_{stall}. On such an occurrence, the presentation would temporarily stop for T_{stall} seconds. Situations like this typically happen if the presentation has been encoded for a bitrate close to the available end-to-end bottleneck bandwidth $\hat{b}$. Furthermore, let P_{drop} denote the probability that a frame is transferred correctly but *dropped* at the receiver because it was received and decoded too late. Finally, let the *probability for presentation breakdown* be denoted by P_{br}.

A summary of the video transport metrics is shown in Table 5. These metrics will be measured using the DVMD (DVB) and an adapted version of the MPEG4IP[6] player (MPEG-4 over IP).

Table 5: CODIS video transport metrics.

Metrics	Symbol	Priority
Frame rate	R_{fr}	Mandatory
Overall bitrate	B_o	Mandatory
Netto bitrate	B_n	Optional
Prob. for packet loss	P_{pl}	Optional
Prob. for pres. stall	P_{stall}	Mandatory
Stall duration	T_{stall}	Mandatory
Prob. for frame drop	P_{drop}	Mandatory
Prob. for pres. breakdown	P_{br}	Optional

5 VIDEO QUALITY

In CODIS videos will be based on MPEG-2 (pure DVB) and MPEG-4 (IP based). Two sources for reducing the presentation quality of digital videos exist: (i) picture

[5] http://lox.csis.gvsu.edu/itl/tools2.htm
[6] http://mpeg4ip.sourceforge.net/

artifacts (impairments) and reduced frame rates stemming from reducing the video data rate in the encoding process, and (ii) picture artifacts and frame losses due to lost or delayed packets.

The ability for a CDN to transport multiple video streams is directly influenced by the video bitrate, thus it is imperative to relate the CODIS transport capacity (number of videos that can be transported concurrently) to the expected end user perceived quality.

Usually, the quality of digital video transmission are measured in terms like block distortion, blurring, edge business, mosquito noise, etc [1, 2]. For CODIS we are also interested into how real human observers would judge the video quality. There are many metrics relating objective quality metrics to subjective ratings [28], many of them, though, have been incorporated into commercial products and can not be used for our experiments [27].

5.1. DVQL-W

The DVQL-W metric [31] has been developed by Rohde & Schwarz and the Institut für Nachrichtentechnik of the TU Braunschweig. Being a special case of an edge detection filter [2], it basically measures block distortion [1] occurring when using high MPEG-2 compression rates, and relates them to previously conducted subjective experiments. It works by summing the squared differences of horizontal neighbor pixels for each column i of the received frame $I(i, j)$ (with size $X \times Y$ pixels)

$$S(i) = \sum_{j=1}^{Y} \left(I(i+1, j) - I(i, j) \right)^2, \; i = 1, \ldots, X-1, \quad (7)$$

yielding characteristic *spikes* at the *block borders*. An overall measure then can be calculated by relating the mean of the $S(i)$ taken at block borders to those inside the blocks.

An advanced version of this metric applies weights to different color channels (either RGB or YUV) and different frame regions, yielding a weighted version called DVQL-W. The digital video quality (DVQ) analyzer of Rohde & Schwarz produces the quality estimate DVQL-W, lying between 0 and 100. The interpretation of this metric is given in Table 6.

An important advantage of this approach is given by the fact that only the received pictures are needed (single ended metrics), no mapping to their originals is required.

Table 6: DVQL-W interpretation.

Interval	Bounds	Interpretation
1	0—20	Bad
2	20—40	Poor
3	40—60	Fair
4	60—80	Good
5	80—100	Excellent

5.2 WOL99

Another metric that correlates highly to a large number of different types of video can be found in [30] relying on special variants of so-called Sobel-filters and HV-filters. Wol99 uses spatio-temporal regions of a video presentation, consisting of several subframes of succeeding frames and will be used for judging MPEG-4 streams sent over IP.

6. BITRATE RULES

In this section we describe preliminary experimental results obtained during the design phase of CODIS.

In order to improve video quality for a given content (characterised by its given spatial and temporal activity) it is necessary to increase the bitrate. Broadcasters and service providers, however, aim at keeping the bitrate low in order to be able to broadcast as many programmes as possible on the same channel. For example a typical satellite channel has 36 MHz bandwidth, which supports transmissions at about 35-40 Mbps [10]. The DVB-T standard allows for data rates up to 30 Mbps, but actually only data rates from 11 to 22 Mbps are used [14]. We conducted experiments to generate rules for finding the minimum bitrate that still allows good quality. This was done by relating video quality to both spatial and temporal activity for a given bitrate. All three values (video quality, spatial activity, temporal activity) were measured by the Rohde & Schwarz DVQ. Input videos were given by the test sequences provided by the video quality experts group (VQEG),[7] encoded at 1.5, 2, 3, 4, 5, 6, 7, 8, 9, 12 and 15 Mbps. The sequences include content with different spatial and temporal activity.

In the following we propose six threshold criteria limiting the measured DVQL-W video quality value, which are used to accept or reject the quality of a sequence. Thus the *lowest bitrate accepted by the criterion* is the best choice for a broadcaster that usually seeks to keep good quality, while still keeping bitrates as low as possible.

The first threshold criterion (criterion A) states that for the whole sequence of frames, the DVQL-W value must not fall below a value of 40, thus quality must be fair, good or excellent in terms of DVQL-W rating (Table 6).

The second criterion (criterion B) states that DVQL-W must not fall below a value of 60, i.e. quality must be good or excellent.

Furthermore there are *soft thresholds*, which means that it is allowed that the DVQL-W value falls below the threshold once in the sequence, but only if the interval does not last longer than one second, which corresponds to two subsequent DVQL-W values, calculated each 400 ms. Therefore a sequence is rejected, as soon as three subsequent DVQL-W values fall below the threshold, or as soon as the DVQL-W value is below the threshold the second time. The period of 1s was chosen, as this is about

[7] http://www.its.bldrdoc.gov/vqeg/downloads.html

the limit for the user's flow of thought to stay uninterrupted [18]. Therefore 1s is about the interval a video sequence may show more artifacts without annoying the user. For the threshold of 40, this is referred to as criterion C, for the threshold of 60 as criterion D. Alternatively the *mean over three subsequent* DVQL-W values may be observed. If the mean falls below the threshold, the bitrate is rejected. Criterion E for a threshold of 40, criterion F for a threshold of 60.

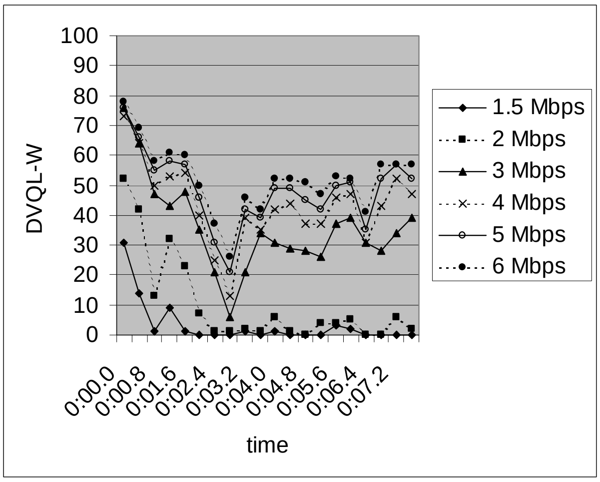

Fig. 4: Video quality for the Rugby sequence (I)

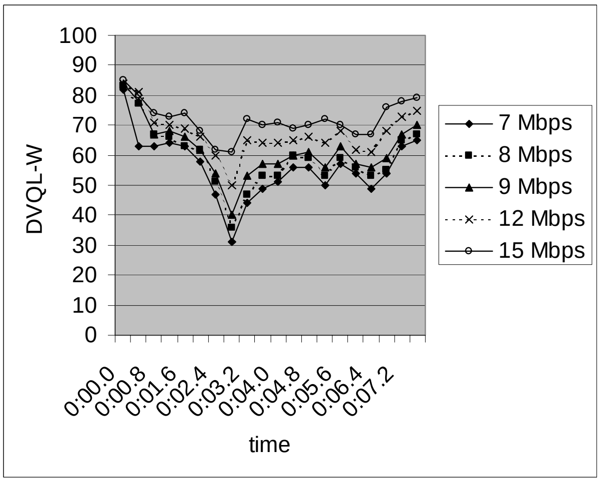

Fig. 5: Video quality for the Rugby sequence (II)

Investigating the Rugby sequence (Fig. 4 and Fig. 5), characterized by a high spatial and temporal activity, for criterion A 9 Mbps is the lowest bitrate not rejected, for criterion B 15 Mbps, for criterion C 6 Mbps, for criterion D 12 Mbps, for criterion E 7 Mbps, and for criterion F 15 Mbps. For each criterion, a map has been generated, showing which bitrate is recommended for which kind of content (Fig. 6).

The used content has also been characterized by a two-dimensional vector containing the amount of spatial and temporal activity the scene shows. For time collapsing the spatial and temporal activity values, for criteria A to D, the mean and alternatively the maximum is used, for criteria E and F, the maximum of all possible averages of eight subsequent values was used, which was found to represent the content best (see example in Table 7).

The question which criterion actually to use was then investigated. For a meaningful criterion, the suggested bitrates should be consistent in the sense that if content X shows a higher spatial or temporal activity than content Y, then a consistent criterion should suggest a bitrate for X which is at least as high as the one suggested for Y.

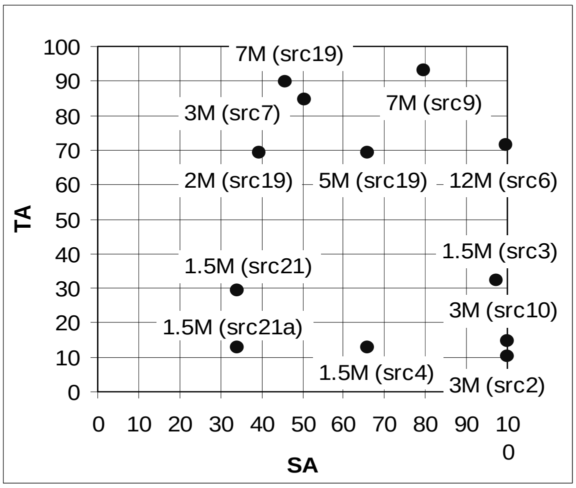

Fig. 6. Required bitrates for all sequences using crit. E

Table 7: Time collapsed spatial (SA) and temporal (TA) activity of src9.

	average	max	max(average8)
SA	67 %	91 %	80 %
TA	77 %	100 %	93 %

The points in Fig. 6 show the spatial and temporal activity of all sequences taken into account, which are labeled with the minimum acceptable bitrate, calculated by applying *criterion E*. The same graphs can be plotted for all other criteria, but criterion E showed the highest consistency.

At low and medium temporal activity levels – even at high spatial activity levels – a bitrate of 1.5 Mbps is sufficient. Only if the maximum spatial activity is reached, an increase in the bitrate is required. At higher temporal activity levels consistency is low and monotony is not granted. Best results are reached with criterion E using the maximum of all averages of eight subsequent values for aggregating spatial and temporal activity values. This aggregation function is a compromise between the maximum and the mean aggregation function, which both do not perform very well themselves. Using the maximum does not seem to be appropriate, because the amount of fluctuations over time cannot be accounted for. For example src21 shows a high temporal activity using the maximum, although only in the end of the sequence high temporal activity occurs and for the rest of the sequence, the temporal activity is much lower. The mean does not seem to be appropriate either, because too much information is lost by averaging. One cannot know if, for example, the mean was a constant value or if high fluctuations around the mean occurred. Therefore criterion E and criterion F using the maximum of all averages of eight subsequent values seem most appropriate.

As criterion E performs better, it is recommended to serve as a hint, which bitrate should be used for compression to reach excellent, good or at least fair video quality:

- For a temporal activity level below 50 % and for spatial activity not reaching the maximum, a bitrate of 1.5 Mbps is sufficient.
- For a temporal activity level below 50 % and a spatial activity of 100 %, a bitrate of at least 3 Mbps is required.
- For temporal activity levels of 50 % to 70 % and a spatial activity not reaching the maximum of 100 %, a bitrate of 2 Mbps is sufficient.
- Above 70 % of temporal activity 3 Mbps to 7 Mbps are required.
- At high temporal activity levels (above 70 %) combined with maximum spatial activity, even 12 Mbps are required.

As can be seen quality and therefore required bitrates are more sensitive to high temporal activity than high spatial activity. Especially at low temporal activity levels, spatial activity does not influence the bitrate very much. Only reaching 100 % spatial activity requires an increase of the encoding bitrate.

7. CONCLUSION

In this paper we have introduced the CODIS framework for measuring the quality of service of a satellite based content delivery network. The framework contains low level transport metrics as well as metrics that can be related to how end users perceive the CDN performance.

We have also introduced rules for broadcasters, yielding a consistent criterion which helps broadcasters to find the optimal bitrate to use for different DVB content.

The project will enter the second phase soon where actual measurements will be performed according to the measure-ment framework.

REFERENCES

[1] ANSI, Digital Transport of Video Teleconferencing/Video Telephony Signals—Performance Terms, Definitions, and Examples, *ANSI Standard T1.801.2-1996*, 1996.

[2] ANSI, Digital Transport of One—Way Video Signals–Parameters for Objective Performance Assessment, *ANSI Standard T1.801.03-1996*, 1996.

[3] ANSI, Multimedia Communications Delay, Synchronization, and Frame Rate, *ANSI Standard T1.801.4-1997*, 1997.

[4] N. Bhatti, R. Friedrich, Web Server Support for Tiered Services, Technical Report HPL-1999-160, HP Laboratories Palo Alto, 1999.

[5] A. Bouch, A. Kuchinsky, N. Bhatti, Quality is in the Eye of the Beholder: Meeting User's Requirements for Internet Quality of Service. Technical Report HPL-2000-4, HP Laboratories Palo Alto, January 2000.

[6] A. Bouch, M. A. Sasse, H. Demeer, Of packets and people: A User-centered Approach to Quality of Service, *Proc. of IWQoS'00*, 2000.

[7] C. J. van den Branden Lambrecht, Color Moving Pictures Quality Metric, *Proc. of ICIP 96*, Lausanne, Switzerland, 1996.

[8] A. B. Downey, Using pathchar to estimate Internet link characteristics, *Proc. SIGCOMM'99*, 1999.

[9] Digital Video Broadcasting (DVB): Measurement Guidelines for DVB Systems, *ETSI TR 101 290 V1.2.1 (2001-05)*, 2001.

[10] G. Fairhurst, DVB Transmission, University of Aberdeen, Dept. of Engeneering, 2001.

[11] Frost & Sullivan, World Caching, Streaming & Multicasting Via Satellites Markets, Report No. 7920-66, Available from: http://www.frost.com.

[12] General quality of service parameters for communication via public data networks, *ITU-T Recommendation X.140*, ITU, 1992.

[13] Methodology for the Subjective Assessment of the Quality of Television Pictures, *ITU Recommendation ITU-R BT.500-10,* 2000.

[14] M. Kühn, J. Antkowiak, Statistischer Multiplex – Was bringt er für DVB-T?, *Fernseh- und Kino-Technik 4*, 2000, 178-183.

[15] K. Lai, M. Baker, Measuring Link Bandwidths Using a Deterministic Model of Packet Delay, *Proc. of ACM SIGCOMM 2000*, August 2000.

[16] M. Liste, Content Delivery Networks (CDNs)—A Reference Guide, Cisco World Magazine, white paper, March 2001.

[17] M. Mathis, M. Allman, A Framework for Defining Empirical Bulk Transfer Capacity Metrics, *RFC 3148*, IETF, July 2001.

[18] M. J. Nadenau, S. Winkler, D. Alleysson, M. Kunt, Human Vision Models for Perceptually Optimized Image Processing—A Review, submitted to *Proc. of the IEEE*, Sep 2000.

[19] J. Nielsen, Usability Engineering, Morgan Kaufman, San Francisco 1994.

[20] V. Paxson, Measurement and Analysis of End-to-End Internet Dynamics, Ph.D. Thesis, Computer Science Division, University of Berkeley, 1997.

[21] V. Paxson, G. Almes, J. Mahdavi, M. Mathis, Framework for IP Performance Metrics, *RFC 2330*, IETF, May 1998.

[22] F. Pereira, Overview of the MPEG-4 Standard, *ISO/IEC JTC1/SC29/WG11 N4668*, March 2002.

[23] R. Rajamony, M. Elnozahy, Measuring Client-Perceived Response Times on the WWW, *Proc. 3rd USENIX Symposium on Internet Technologies and Systems (USITS)*, San Francisco, 2001.

[24] U. Reimers, Digital Video Broadcasting (DVB*). The international standard for digital television*. Springer-Verlag Berlin Heidelberg, 2000.

[25] N.B. Seitz, S. Wolf, S. Voran, R. Bloomfield, User-Oriented Measures of Telecommunication Quality, *IEEE Communications Magazine 1*, 1994, 56-66.

[26] Tektronix, A Guide to Maintaining Video Quality of Service for Digital Television Programs, 2000.

[27] Video Quality Experts Group, Final Report from the Video Quality Experts Group on the Validation of Objective Models of Video Quality Assessment, March 2000.

[28] Zhou Wang, Objective Image/Video Quality Measurement—A Literature Survey, Available: http://www.ece.utexas.edu/~bevans/ courses/ee381k/projects/fall98/zhou/literatureSurvey.pdf

[29] A. Watson, M. A. Sasse, Evaluating Audio and Video Quality in Low-Cost Multimedia Conferencing Systems, *Interacting with Computers 8*(3), 1996, 255-275.

[30] S. Wolf, M. H. Pinson, Spatial-Temporal Distortion Metrics for In-Service Quality Monitoring of Any Digital Video System, *Proc. of SPIE International Symposium on Voice, Video, and Data Communications*, Boston, MA, September 11-22, 1999.

[31] A. Wörner, DVQ—An objective picture quality measurement not requiring a reference (NR), Committee T1–Telecommunications Committee Group T1A1.1, T1A1.1/2000-044, Charleston, 2000.

[32] Zona Research Inc., The Economic Impacts of Unacceptable Web-Site Download Speeds, April 1999.