Enriched Network-aware Video Services over Internet Overlay Networks

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Deliverable D5.3

Final Specification of Metadata Management, Dynamic Content Generation and Adaptation, Adaptation and Caching Node Functions

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Abstract: This deliverable presents work achieved in work package 5 (WP5) during the third year of the ENVISION project. We present novel resilient multi-channel scheduling algorithms for layered streaming to improve the video quality and reduce the packet loss and the late arrival packets. Then, we propose an efficient overlay construction algorithm. This algorithm aims to ensure good layers propagation in the network and reduction of the quality bottleneck. In addition, we propose peers' quality level aware bandwidth allocation scheme inspired by the auction mechanisms in order to achieve better exploitation of the available bandwidth. In order to evaluate the proposed content delivery and adaptation mechanisms, we studied the state of the art of the subjective assessment mechanisms and the scenarios of their use in the context of ENVISION. Finally, we present ENVISION long term caching architecture and the streaming client software architecture.

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EXECUTIVE SUMMARY

In continuity with D5.2, this deliverable describes proposed mechanisms, techniques and algorithms for content generation and adaptation that support the delivery of multimedia content to a large number and heterogeneous end-users. It focuses on the use of layered streaming in P2P network supported by ENVISION architecture. This deliverable presents the following contributions:

- Different layered-video packet multi-channel scheduling algorithms, their evaluation and comparison.
- An overlay construction algorithm, aware of the peers capacities and preferences, in order to ensure a large peers quality satisfaction.
- A bandwidth allocation mechanism for P2P layered streaming, which is also aware of the peers requirements in terms of quality level.
- A detailed state of the art about subjective quality assessment and scenarios of its use to assess the end user quality satisfaction in ENVISION.
- The architecture of ENVISION long term caching and its interaction with other components via CINA interface.
- An overview of the streaming client software architecture.

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1. INTRODUCTION

This deliverable contains the main achievements of work performed in work package 5 during the third year of the project. It presents the final specification of the dynamic content generation, adaptation and caching functionalities.

After exploring the benefits of using multiple links to distribute the content in D5.2, in this deliverable we propose several layered-video packet multi-channel scheduling algorithms. We focus on solutions appropriate to wireless/mobile/sensor nodes and having a simple decision function, low computation afford, small state storage requirements and little channel feedback information.

We tackle after that the problem of quality bottleneck, where some peers having the capacity to obtain high quality, but due to the overlay structure, they are connected to peers having only low quality layers. To deal with this problem, we propose in this deliverable a new overlay construction approach based on preferential attachment probability for building an efficient and churn-tolerant overlay for layered video delivery.

Once the overlay is constructed, another point should be considered, namely the bandwidth allocation. Indeed, a bad distribution of the upstream peers' bandwidth can lead to underutilisation of the available bandwidth even with a powerful scheduling algorithm. This is why we propose a new bandwidth allocation algorithm for layered streaming, where we model the competition on the bandwidth as an auction game. In addition to experimental results that will be detailed in D 6.3, we prove theoretically the convergence of the proposed solution to Nash equilibrium.

In this deliverable and D5.2, we discussed several complexities involved in layered streaming over P2P networks due to bandwidth fluctuation, peer churn and peer's unreliability. We proposed smoothing mechanism, efficient overlay construction and bandwidth allocation mechanisms. The main goal is to ensure a good perceived quality for the end user. One method to evaluate this quality can be performed to assess the quality of the stream with respect to viewer perspective. In this deliverable we present a detailed state of the art of the quality subjective measurement and we explore the possible assessment scenarios that can be envisaged for ENVISION.

Finally we detail the architecture of the ENVISION cache, its integration in the global architecture of ENVISION and its interaction with other ENVISION components via the CINA interface.

2. SVC CONTENT SCHEDULING OVER MULTILINK

***Objectives*—**we propose and evaluate several layered-video packet multi-channel scheduling algorithms. The scheduler's properties are suitable for wireless/mobile/sensor nodes and have a simple decision function, low computation afford, small state storage requirements and little channel feedback information requirements. Moreover we are extending the algorithm provided for SVC smoothness in ENVISION D5.2 deliverable [D5.2] to the multi-link case.

2.1 Background & Challenges

Mobile networks are formed by a large number of small nodes communicating over wireless links. These nodes have exponentially growing transmission range, processing, storage and energy resources capabilities and are injecting more and more video content into the network, this trend being line with the recent growth of user generated content. However the network resources are limited and congested, and as a result unreliable and unstable, thus streaming video solutions for wireless networks must ensure video quality under these conditions as has been realised e.g. in [GGG11][PBR11][GTE07].

A video streaming application encodes, packetises and transmits video frames in real-time. That is, every streaming video frame needs to meet a play-out deadline. Currently, most of the networks support real-time services only in a best effort manner. Therefore, video streaming services have to include special measures to be resilient against packet loss and late arrival. Over the last decade streaming over multi-channels (also called multi-path, or networks) has been suggested to improve the video quality over the Internet [WWT09] [JF08] [AWT02] [GLT02] [LSG01] [NA04], in Peer-to-Peer (P2P) networks [WXR10] [AR05] [GLL08] [ZXZ09] and over wireless ad-hoc networks [SZG04] [MLP03]. Multi-channel video transmission is often coupled with adaptive/scalable layered-video encoding (H.264/SVC) to overcome channel rate variation and heterogeneous video client capabilities. Using multiple channels in the layered-video transmission has led to new challenges such as video packet scheduling and new multi-channel encoding schemes [MLP03] [RKP11] [ZWT10] [ZGZ09]

2.2 Scientific Work

In this study, we suggest and evaluate several algorithms for on-line scheduling of layered-video content into multiple channels. These algorithms were developed specifically for wireless/mobile/sensor nodes because they have the following properties:

- Simple decision function
- Low computation afford
- Small state and storage requirements
- Little channel feedback information

Considerable attention in the literature has been paid to scheduling of multiple video streams, in particular over P2P networks [AR05] [GLL08] [ZXZ09]. These works are based on the assumption that many peers have the video content, and the video client (receiver) can generate requests for video packets from many peers. In most cases the success of a video streaming session depends primarily on the number of replications of the video content in the network. However, nodes in wireless network have limited storage. Thus, it cannot be assumed that a video content captured in one node can be widely replicated in the network. In fact, we assume that each video has one source, which is also the case for live content source.

The authors in [AWT02] proposed an optimal scheduling strategy to minimise the overall video distortion, but their approach is closely related to Multiple Description (MD) coding, which is less efficient than layered coding. The algorithm proposed in [RR06] minimises the base layer losses, but it assumes that the base layer rate is equal to the enhancement. This video model is fairly idealised and can only be approximated by fine grained scalability (FGS). In [BSW07] a scheduler was described to maximise video quality by prioritising the most important packets. A similar strategy was discussed in [RKP11], in which the channels are prioritised according to their packet loss rate and the important packets are transmitted over the high priority channels. However, it is predicated on a fixed channel error-rate over time.

As plotted in Figure 1, the packet error rate changes dramatically over time. Our proposed schedulers overcome this drawback by extending this strategy to time varying channels. These new schedulers are compared to the fixed strategy and the optimal off-line scheduler.

We implemented the schedulers and compared their performance on a real data trace. The experimental results confirm that our short-term scheduler outperforms the other schedulers. In particular, it reduces the error cost by 14%-24% comparing to the fixed scheduler suggested in [RKP11] [BSW07]. Furthermore, its performance is very close (~1% difference) to the off-line scheduler (the best possible assignment).

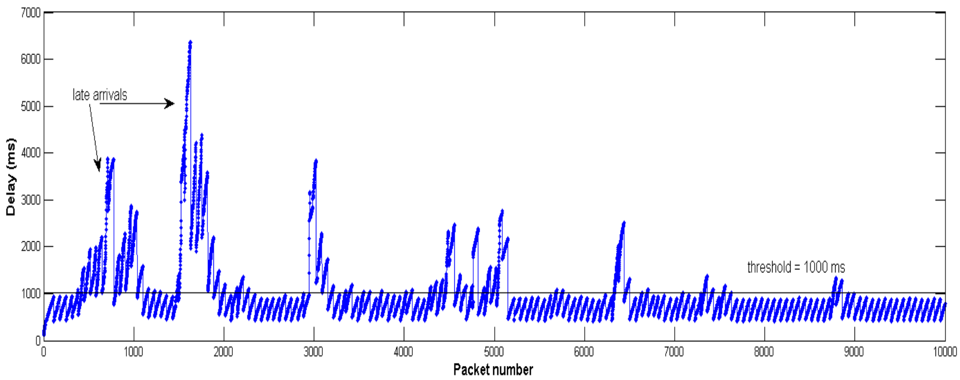

Figure 1: Example of measured delay in video streaming

In this section we provide an overview of the multi-channel video transmission system under consideration. As shown in Figure 2, the system consists of three parts: the video server, the M multiple channels and the video client. The video server is a mobile/wireless/sensor node that consists of a video source, a video encoder, and a module for stream splitting and channel protection. The video client is a mobile/wireless/sensor node that consists of a module for joining and decoding the channel protection, a video decoder and a viewer. These components are described briefly in the following paragraphs.

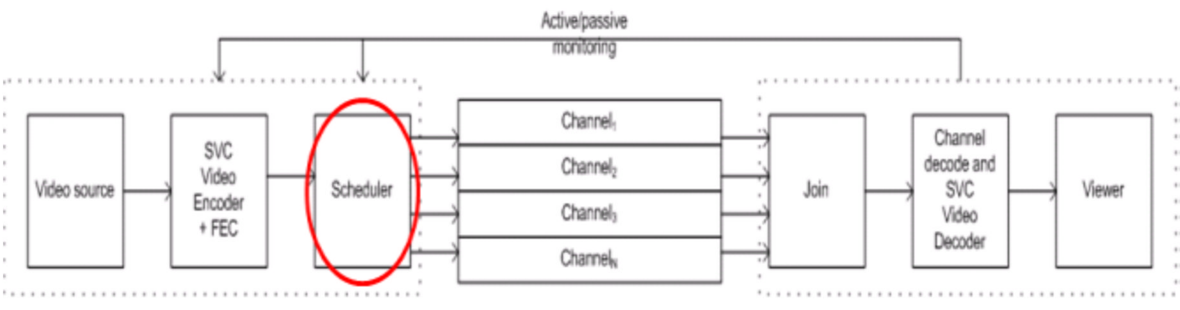

Figure 2: System under Consideration

We assume that a space-time discrete video signal is used as input to the layered video encoder, which is characterised by its operational distortion-rate (DR) function D_{enc}. After source coding, the compressed layered video stream is prepared for transmission by the channel codec. This involves packetisation and forward error control (FEC) combined with interleaving to reduce the effect of burst errors. After channel encoding, the video layers are scheduled to channels and then the video packets are transmitted over the M channels according to their layer-to-channel mapping.

For multi-channel video streaming applications, video packets are transmitted over the multiple channels and need to meet a play-out deadline. Decoded video quality at the receiver end is therefore affected by two factors: encoder compression performance and distortion due to packet loss or late arrivals. In this paper we focus on the video distortion from packet loss and late arrivals.

1) $D_{dec} = D_{enc} + D_{loss}$

   We assume that D_{loss} is linearly related to the packet loss and late arrival rate:

2) $D_{loss} = a \cdot (P_{loss} + P_{late})$ (2)

   For layered video, the video distortion due to a packet loss and late arrival depends on the packet layer. Losing a packet from the basic layer is the worst and results in high video distortion. That is,

3) $D_{dec} = D_{enc} + \Sigma \text{layers} (D_{loss} (\text{layer}))$ (3)

   We assume that for each video layer *i* a constant packet lost cost C_i is given, where $C_1 > C_2 > \cdots > C_M$. The parameters C_i, i = {1, …, M}, represent the penalty on loss or late arrival of a single video packet in layer i.

$$Cost = \sum_i^n C_i(S_i - R_i)$$

Equation 1: Cost calculation

Where S_i represent the number of sent packets in link i, and R_i represent the number of on time received packets in link i.

2.2.1 SVC Scheduler over Multilink

In this section, we describe and analyse the new scheduling algorithms for the multi-channel video system. In addition, we describe the optimal off-line scheduler and revise the fixed channel scheduler in [RKP11]. The following procedures define the schedulers. In this description we refer to both loss packets and late arrivals as lost packets.

- The fixed priority scheduler [RKP11] calculates the channel priority once – in the second interval. The priorities are driven from the loss rate (including late arrivals) reported in the first time interval. These channel priorities are used until the end of the session.
- The long-term scheduler calculates the channel priorities for each time interval. The calculation is based on the average loss rate from the beginning of the session until the last interval. For each time interval the mapping is based on the new calculation on the channel results. According to the model, the average loss rate is the stationary state probability P(B) of state B (bad). We assume that in state G (good) all packet transmission arrive on time while in state B all packet transmission are either lost or late arrive. Thus, the error rate in this model is equal to the

stationary state probability P(B) of state B (bad). This scheduler calculation is based on estimation of each channel parameters PGB and PBG according to the channel quality from the beginning of the session until the last interval, and prediction of the channel quality in the next time interval according to the expected channel loss rate.

- The short-term scheduler calculates the channel priorities for each time interval. The calculation is based on the loss rate in the previous time interval. In each time interval the mapping is based on the new calculation on the channel results. However, it uses the statistics of the previous time interval solely.
- The weighted scheduler is a hybrid of the short and the long term schedulers. Its decision is based on the long-term average of the channel packet loss and the report of the previous interval.
- The off-line scheduler simply chooses the best possible assignment by setting the channel priorities in time interval *t* according to the channel performance at the same time interval. Obviously, it cannot be implemented in a real system. We use it solely as a performance reference.

It should be noted that all these schedulers have the list of features discussed in the introduction. They are very simple to compute, with low computational afford. Storage needs for calculation are low. Furthermore, they only use small feedback messages with the statistic of loss rate of each channel in the previous time interval.

2.2.2 Scheduling SVC packets in a multi-link environment

In the previous section, we investigated the best approach to calculate the channel's error rate, based on the previous time intervals, and tried to state how many past intervals should be considered in order to provide the best video quality.

Once we know how many time intervals are required we can use that information in order to develop a scheduling algorithm for SVC layered stream which will provide us with minimal quality distortions.

The following algorithm, described in Figure 3, extends the work which was described in D5.2 on SVC scheduling to a multi-channel environment. The goal is to determine which packet should be transmitted on which channel at a given moment. The x-axis is the time since a packet is generated; the y-axis is the priority line where lowest values have higher priority. The blue squares represent SVC packets/ NALs as shown they are inserted to lines $Line_1$ – $Line_N$ with accordance to their priority, in a way that L1 is the base layer and Ln is the least important layer. Packets are progressing from left to right with accordance to their timestamp, where the scheduler is aiming to deliver packets over modems to guarantee their deliver on time and not arriving too late for display. Thus the dotted lines for the Wi-Fi and 3G access are positioned with correspondence to each channel's delay (as we learned from the monitoring process) in a way that if a packet has passed (left to right) the modem/link line it will arrive late with a high probability. The final vertical line to the right is the display time at the receiver, its positioned is determined with respect to the service allowed delay.

Since y-axis is the inverse of the priority, lower positioned packets should receive higher priorities and in this algorithm the scheduler shall always transmit the lowest packet in the system buffers. Thus packets are transmitted according to their time-priority nature depending on the line curves $Line_1$ - $Line_N$. The line slope and gaps between lines influence the aggressiveness factor of the scheduler, as defined in D5.2 (aggressiveness versus conservative approaches). Steep lines or small gaps between them describe an aggressive scheduler, where higher layers are transmitted before all the more important layer packets are delivered.

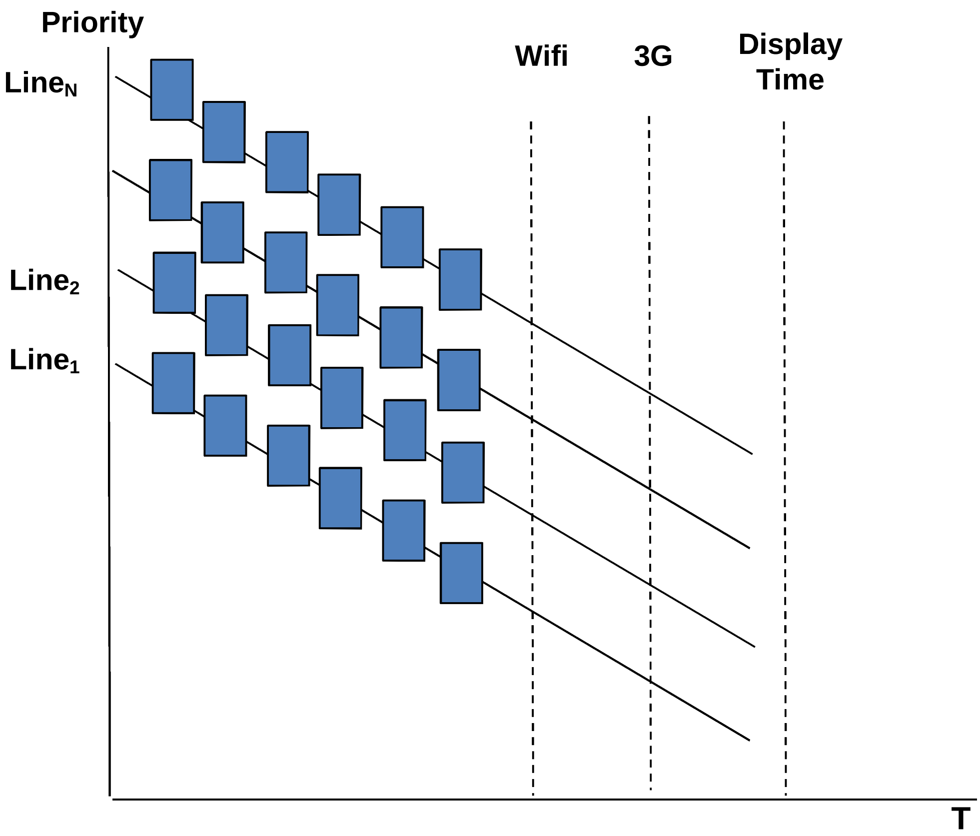

Figure 3 : Proposed Multilink priority scheduler

2.2.2.1 Assessments and simulations

In the following section we describe the on-going assessment through simulation, dealing with combinations of aggressive and conservative transmission methods. We are able to control the aggressiveness factor using both the lines slope and the distance between the lines. Steeper slopes indicate an increased aggressiveness factor, and so do closer lines.

The following graphs in Figure 4 describe the different options under consideration in the simulation work.

In the top right graph, we play with the curves slopes and the gaps between curves to control the conservative versus aggressiveness of the scheduler.

In the top left graph, and in the bottom left graph we play with the gaps/slopes in an un-equal manner between layers, to achieve an approach which is conservative with the base layer and more and more aggressive with the higher enhancements layers.

In the bottom left graph, we change the slope over time, where we can achieve a conservative scheduler when packets are sent close to the limits of modems delay, and a more aggressive approach/slope when packets are sent far from the modem delay limit.

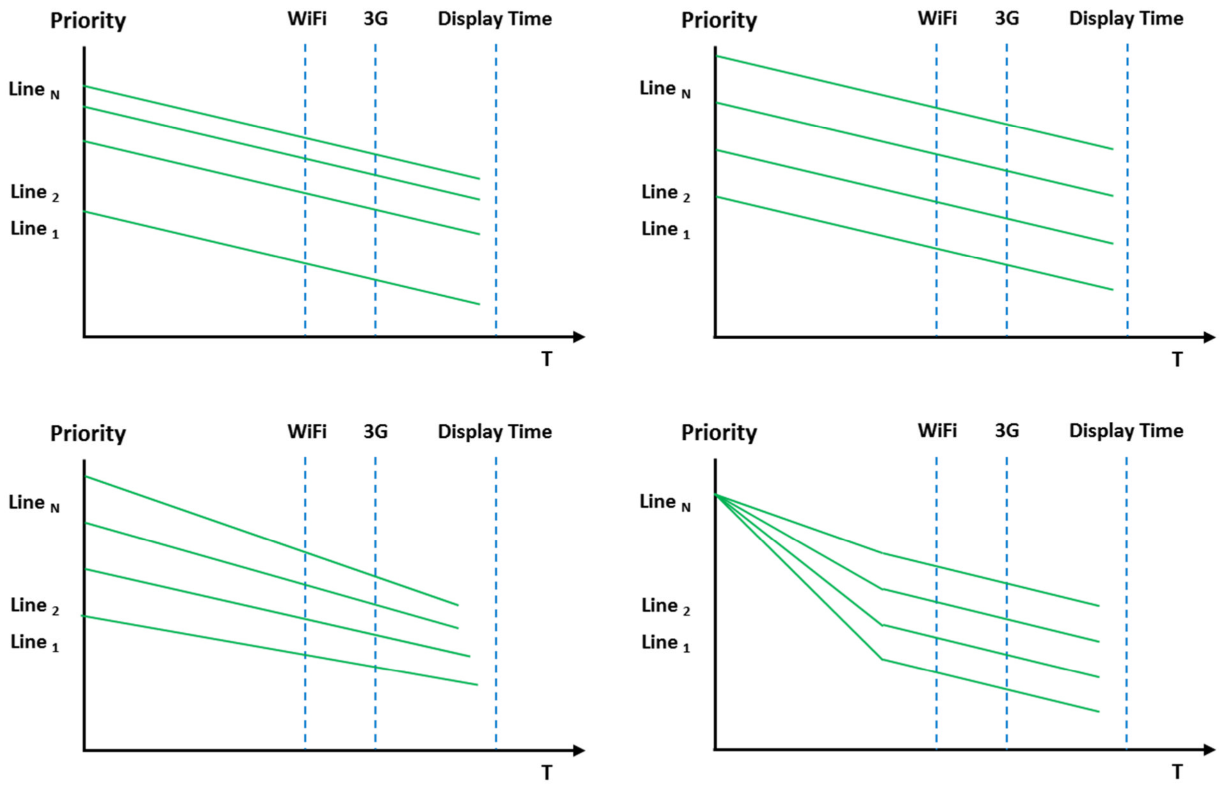

Figure 4: Scheduler options under consideration

2.3 Results

In this section we describe the trace collection and the results of the schedulers with respect to section 2.2.1. To validate the scheduler's performance we collected a real data trace. First, we transmitted video streams using several channels under various conditions. The transmission of the channels used LU60 of LiveU [AR05], and employed between one to five modems connected to three different networks. Each modem had a different connection to the internet. We recorded the received data with LiveU's server (LU1000) and also using 'Wireshark' software. For each packet in each transmission from each channel, we recorded the packet sequence number, the transmit time stamp and the received time stamp.

The recordings were made throughout the day including both peak (busy hours) and off-peak hours. Each experiment consisted of 5 samples of video transmissions using one to five simultaneous channels. The experiment was repeated 10 times with long video files (about 15 minutes). In addition, the experiment was repeated twice with short video files (five minutes) to test whether the observed statistic behaviour also fit short transmissions. Overall, the recording trace included statistics for more than 12 million packets.

Figure 5 presents the scheduler cost calculations for a time interval of 1 second. The fixed scheduler and the long-term scheduler exhibited similar performance. It can be seen that the short-term scheduler outperformed both of them and achieved results that were very close to the off-line scheduler (the best possible assignment). Additional results regarding the weighted schedulers are presented in Figure 6. It can be seen that increasing the value of the parameter α (where α is just a linear interpolation factor between long and short term) results in cost reduction. In other words, because of the rapid changes in channel conditions over time, it is better to consider only the last statistic and ignore the channel long-term history.

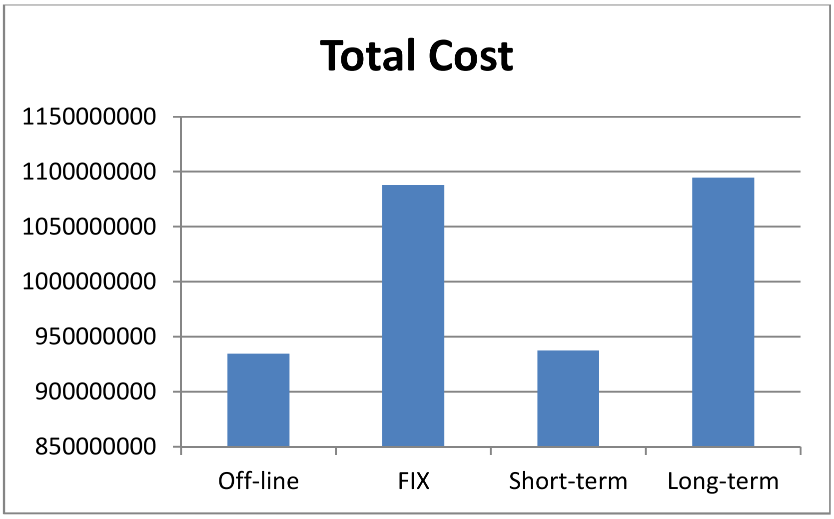

Figure 5: Scheduler Costs

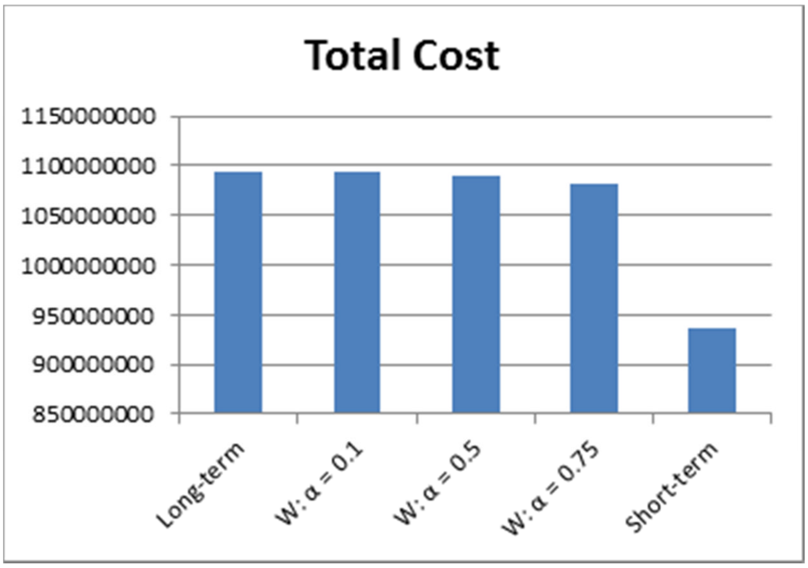

Figure 6: Time Weighted Scheduler Costs

2.4 Dissemination and Exploitation potential

In this study we suggest several scheduling algorithms for multi-channel layered-video transmission over wireless networks. The schedulers are specifically designed for mobile nodes as they are very simple to compute and have low computation effort. They require small storage for the calculation and they only use small feedback messages to provide the loss rate statistic for each channel in the previous time interval.

Our empirical results show that the short-term scheduler outperforms the other schedulers. In particular, it reduces the error cost by 14%-24% compared to the fixed scheduler suggested in [RKP11] [BSW07]. Furthermore, its performance is very close (~1% difference) to the off-line scheduler (the best possible assignment). This work was accepted to [NA12] and the results of this work are exploitable and planned to be integrated into LiveU LU60/40 product lines.

3. QUALITY BOTTLENECK IN P2P LAYERED STREAMING

In P2P streaming systems, peers may join and leave the system without prior notice; this behaviour is called "churn". Churn may be a great problem for certain P2P applications that have strong delay constraints such as video streaming, IP telephony, online video games, etc. Regarding the peers organisation, peers that undergo churn need to quickly recover by fetching new neighbours, and establishing new connections with them. This behaviour may lead to severe delay issues and service discontinuity.

In the case of P2P video streaming applications, an aspect of the churn has been overcome by using video content playback that supports the lack of partial data. Examples of such codecs are MDC [GVK01] and SVC for video content. With such codecs, the playback of a video content does not severely suffer from neighbour departure, if the leaving neighbour was providing supplemental data. However, the quality of the received content will be traded for the service continuity. By using layered video content, churn can be tolerated to a certain degree [PWC03]. However, when decomposing the stream, in case of SVC, to incremental and dependent parts, other issues may arise [ZYR09].

Heterogeneous peers are now able to consume a tailored content that meets their different capabilities and preferences. This feature will make the overlay somehow similar to a multi-overlay where the exchanged contents are dependent. Peers are now bothered with the version (or the quality level) they are requesting. Such issue can be overcome by using an efficient scheduling [ZYR09] [RA03] [XYY09] [CYY08] of the data chunks from the neighbouring peers.

However, the availability of the desired data chunks in the neighbourhood depends on the overlay formation. If an arriving peer is connected to peers having only a low quality, this peer will be constrained to request this quality, and share it with other peers even if its capabilities and/or preferences can permit him to obtain a higher one.

Indeed, overlay formation becomes the critical component in the design of a layered video P2P streaming system. Since peers contribute to the overlay by the consumed content or less, peers need to be carefully connected to other peers in order to avoid what we call the "quality bottleneck". If peers' characteristics are not considered while building the overlay, more capable peers may be pushed to the edge of the overlay and vice versa, less capable peers be placed in the core of the overlay in such a way that only a low quality is circulated from the source to the other peers.

Finally, churn will make the overlay formation more challenging, since a peer may see its downstream decreasing, and so its contribution to the overlay will decrease in consequence.

4. BANDWIDTH ALLOCATION FOR LAYERED STREAMING

Layered streaming, such as Scalable Video Coding (SVC), provides a convenient way to perform video quality adaptation to adjust to the changing network conditions and receiver preferences. A layered stream consists of a base layer and multiple enhancement layers. Receivers can adjust the video quality level to their capability by subscribing to different number of layers using pulling distribution approach. In a P2P network, it is natural to request the layer from different peers (upstream peers). Thus, each upstream peer shares its upload bandwidth among different peers to serve different layers. Finding a solution to resolve bandwidth conflicts among peers in order to maximise benefits of both upstream and downstream peers while respecting the layers importance, their dependencies and the peers' priorities is highly challenging in P2P networks. The following section describes the stat of the art for bandwidth allocation in layered streaming.

5. QUALITY ASSESSEMENT

5.1 Introduction

The popularity of digital video content together with the increasing number of connected end-users through heterogeneous networks motivates the utilisation of Peer-to-Peer (P2P) networks. In these networks, the ability to provide a high level of user experience is a challenging task. This task is supported by effective evaluation of service quality. In order to realise such a quality evaluation service, an assessment model is required to bridge relevant influencing factors with end users' experience.

The typical approaches to video quality assessment are mostly based on Quality of Service (QoS). QoS is referred as a performance measure from the perspective of system performance or reliability. However, QoS metrics do not take into account human users' perception, therefore there is a lack of correlation between QoS and users' opinions. The concept of Quality of Experience (QoE) has been introduced [R04] to fill this gap between service performance and user experience. QoE is commonly defined as the overall acceptability of a service as subjective perceived by end users. Using models based on the concept of QoE, it is possible to measure the extent to which a user-centric service (e.g. video content distribution) achieves its objectives.

The video quality assessment techniques can be categorised as objective and subjective video quality assessment. The following sub-sections will provide a brief state of the art for objective and subjective video quality assessment techniques.

5.2 Objective Quality Measurement

In this section, we highlight the techniques for objective quality measurements.

The Structural Similarity Index (SSIM) [ZLAC04] is a top-bottom approach using a model of the Human Visual System [WS05]. The Perceptual Evaluation of Video Quality (PEVQ) is use to estimate video quality degradation introduced in the content networks. The basic principle behind this approach is spatial and temporal measurements. The output of PEVQ is Mean Opinion Score (MOS) [ITUT96] value ranging from 1 (bad) to 5 (excellent) as well as additional indicators for more detailed analysis.

Video Quality Metric (VQM) measures the perceptual effects of video impairments such as blurring, global noise, block distortion and colour distortion. These different effects of video impairments are then combined into one single metric [SMH99].

Moving Picture Quality Metric (MPQM) is an objective quality metric for moving pictures using a vision modelling approach [CJO96]. V-Factor is a specific MPQM implementation specifically designed for the IPTV), and incorporates MPQM research that several labs have developed over the last years [CMP09].

There are also different metrics that are established such as performance and feasibility, to evaluate the design of objective quality assessment solutions. Considerable effort has been made to improve the correlation between the estimations given by assessment models and actual human users' opinions [WS07]. The fitness of the objective evaluation results to the corresponding user scores is considered as the performance of an objective assessment model. The performance of a model can be measured with various metrics according to the statistical requirements. In order to assess and compare the performance of different models in a quantitative manner, tests must be executed in a controlled environment under identical conditions for all target models. Very complex resource-intensive algorithms and measurement methods are commonly adopted aiming at increasing the performance of quality evaluation. However, such sophisticated model can also complicate an evaluation process. Therefore feasibility is considered when benchmarking an objective model to evaluate the difficulty of applying the model to accomplish a designated evaluation task. Feasibility is a multi-faceted qualitative concept encompassing the amount of information required, the computational intensity and practicality of deployment. Feasibility determines whether or not an objective model is suitable to be utilised in specific assessment scenarios to provide an expected level of performance [PMS08].

5.3 Subjective Quality Measurement

The "subjective quality assessment", consists of the use of human observers who should score video quality during experiments called "quality assessment tests". Since video quality is a subjective notion therefore subjective quality assessment can be considered as one of the method to measure video quality. However, many recommendations have to be followed when realising subjective quality assessment tests, otherwise the results of these tests often become useless (because of the lack of precision on the quality scores obtained during these tests). A wide variety of basic test methods have been used in television assessments. In practice, however, particular methods should be used to address particular assessment problems. A survey of typical assessment methods recommended by ITU is provided below [ITU02].

5.3.1 The double-stimulus impairment scale (DSIS) method (the EBU method)

In this method videos are shown consequently in pairs. This method is also referred as Degradation Category Rating (DCR). The first one is the reference, and expert is informed about it while the second one is impaired. After their playback, expert is asked to give his opinion on the second one, keeping in mind the first one. During different sessions (each session last up to half an hour) the assessors are shown series of pictures or sequences in random order and impairments covering all the essential combinations. It is important to note that the reference (unimpaired) picture is included in the pictures or sequences to be assessed. Finally, the mean score is calculated at the end of the session for each test condition and test picture.

When low-resolution picture formats are used (e.g. CIF, QCIF), it could be useful to display the reference and the test sequence simultaneously on the same monitor.

The following five-level scale for rating the impairment should be used:

5. Imperceptible
4. Perceptible but not annoying
3. Slightly annoying
2. Annoying
1. Very Annoying

5.3.2 The double–stimulus continuous quality-scale (DSCQS) method

In this method, the assessor evaluates a pair of pictures, having the same source but one via the process under examination and the other one directly from the source. Thus, the test sequences are presented in pairs, consisting of the same sequence being presented first through one system under test and then through another system. The Double Stimulus Continuous Quality Scale (DSCQS) method is widely used for the quality assessment of systems for television broadcasts. This method is extremely useful in cases where the full range of quality conditions can’t be presented.

The system under tests (I, J, K, etc.) are generally combined in all possible n (n-1) combinations like IJ, JK, KI, etc. Thus, all the pairs of sequences should be displayed in both the possible orders (e.g. IJ, JI). After each pair a judgment is made on which element in a pair is preferred in the context of the test scenario.

Regarding the time pattern for the stimulus presentation, if the voting time should be less or equal to T seconds depending on the mechanism used, then the presentation time should be about T seconds, and it may be reduced or increased according to the content of test material.

There are certain variations of the PC methods that utilise a categorical scale to further measure the difference between the pair of sequences.

5.3.3 Single-stimulus (SS) methods

In this method, a single image or group of images are presented and the expert/assessor provides an index of the complete presentation. This method is also referred as absolute category rating (ACR). The method specifies that after each presentation the subjects are asked to evaluate the quality of the sequence shown. Regarding the time pattern for the stimulus presentation, if the voting time should be less or equal to T seconds depending on the mechanism used, then the presentation time should be about T seconds, and it may be reduced or increased according to the content of test material. Normally, the five-level scale is used for evaluation however a nine level scale may be used if higher discriminative power is required.

There are three types of single stimulus methods used in video assessments.

5.3.3.1 Adjectival categorical judgement methods

In adjectival categorical judgements, an image or image sequence are assigned to one category (among different set of category). The category may reflect judgements of whether or not an attribute is detected (e.g. to establish the impairment threshold). The categorical scale that assesses the image quality and impairment are used as shown in Table 5. Several other scales are also used to assess the text legibility, reading effort, and image usefulness.

Quality	Impairment
5 Excellent	5 Imperceptible
4 Good	4 Perceptible, but not annoying
3 Fair	3 Slightly annoying
2 Poor	2 Annoying
1 Bad	1 Very annoying

Table 1: Quality and Impairment Scale

5.3.3.2 Numerical categorical judgement methods

5.3.3.3 Non-Categorical judgement methods

In this method, a value is assigned to each image or image sequence. There are multiple variations of this method. In continuous scaling, the assessor assigns each image or image sequence to a point on a line drawn between two semantic labels.

In numerical scaling, the assessor assigns each image or image sequence a number that reflects its judged level on a specified dimension (e.g. image sharpness). The range of the numbers used may be restricted (e.g. 0-100).

5.3.4 Stimulus-comparison methods

In stimulus-comparison methods, two images or sequences of images are displayed and the assessor provides an index of the relation among the two presentations. The image sequences used are generated in the way similar to SS methods. The resulting image sequences are then combined to form the pairs that are used in assessment trials.

There are multiple types of stimulus-comparison methods that are widely used in video assessments.

5.3.4.1 Adjectival categorical judgement methods

In adjectival categorical judgement methods, observers assign the relation between members of a pair to one of a set of categories that, typically, are defined in semantic terms. These categories may report the existence of perceptible differences (e.g. SAME, DIFFERENT), the existence and direction of perceptible differences (e.g. LESS, SAME, MORE), or judgements of extent and direction. The comparison scale is shown in the table below (Table 6).

-3	Much worse
-2	Worse
-1	Slightly worse
0	The same
+1	Slightly better
+2	Better
+3	Much Better

Table 2: Quality Scale

This method yields a distribution of judgements across scale categories for each condition pair. The way that responses are analysed depends on the judgement made (e.g. difference) and the information required (e.g. Just-noticeable differences, ranks of conditions, “distances” among conditions, etc.)

5.3.4.2 Non-categorical judgement methods

In non-categorical judgements, observers assign a value to the relation between the members of an assessment pair. There are two forms of this method:

In continuous scaling, the assessor assigns each relation to a point on a line drawn between two labels.

Scales may include additional reference labels at intermediate points. The distance from one end of the line is taken as the value for each condition pair.

In the second form, the assessor assigns each relation a number that reflects its judged level on a specified dimension (e.g. difference in quality). The range of numbers used may be constrained or not. The number assigned may describe the relation in “absolute” terms or in terms of that in a “standard” pair. Both forms result in a distribution of values for each pair of conditions. The method of analysis depends on the nature of the judgement and the information required.

5.3.5 Single stimulus continuous quality evaluation (SSCQE)

The introduction of digital television compression will produce impairments to the picture quality which are scene-dependent and time-varying. Even within short extracts of digitally-coded video, the quality can fluctuate quite widely depending on scene content, and impairments may be very short-lived. Conventional ITU-R methodologies alone are not sufficient to assess this type of material. Furthermore, the double stimulus method of laboratory testing does not replicate the SS home viewing conditions. It was considered useful, therefore, for the subjective quality of digitally-coded video to be measured continuously, with subjects viewing the material once, without a source reference.

5.3.6 Simultaneous double stimulus for continuous evaluation method (SDSCE)

The idea of a continuous evaluation came to ITU-R because the previous methods presented some inadequacies to the video quality measurement of digital compression schemes. The main drawbacks of the previous standardised methods are linked to the occurrence of context-related artefacts in the displayed digital images. In the previous protocols, the viewing time duration of video sequences under evaluation is generally limited to 10 s which is obviously not enough for the observer to have a representative judgement of what could happen in the real service. Digital artefacts are strongly dependent upon the spatial and temporal content of the source image. This is true for the compression schemes but also concerning the error resilience behaviour of digital transmission systems. With the previous standardised methods it was very difficult to choose representative video sequences, or at least to evaluate their representativeness. For this reason ITU-R introduced the SSCQE method that is able to measure video quality on longer sequences, representative of video contents and error statistics. In order to reproduce viewing conditions that are as close as possible to real situations, no references are used in SSCQE. When fidelity has to be evaluated, reference conditions must be introduced. SDSCE has been developed starting from the SSCQE, by making slight deviations concerning the way of presenting the images to the subjects and concerning the rating scale. The method was proposed to MPEG to evaluate error robustness at very low bit rate, but it can be suitably applied to all those cases where fidelity of visual information affected by time-varying degradation has to be evaluated.

5.4 Subjective Measurement for ENVISION

In deliverable D5.2, we discussed the several complexities involved in layered streaming over P2P networks due to bandwidth fluctuation and peer's unreliability. In this case, the correct decision regarding the selection and the scheduling of the layers is crucial: how many layers to request, in which order and from which peer. We proposed amplitude and frequency reduction to ensure the smooth quality of the stream in deliverable D5.2. The subjective evaluation can be performed to assess the quality of the stream with respect to viewer perspective.

Following possible assessment scenarios can be envisaged for ENVISION.

The amplitude reduction mechanism aims to reduce the transition from the higher layer to very lower layer (amplitude reduction) and vice versa. The video quality assessment for amplitude reduction can be performed by examining the optimal trade-off between the temporal and SNR scalability. The video basement method such as double stimulus continuous quality scale (DSCQS) can be used with different frame rate and quantisation parameter but having fixed spatial resolution. The experiment can be performed for different content type such as content having high motion, low motion etc.

Similarly, the subjective evaluation can be performed for frequency reduction mechanism. The objective of frequency reduction is to reduce the number of changes from one layer to another that affects user's perception of stream as described in deliverable D5.2. Several other possible scenarios can be evaluated depending on the content type. These scenarios can be based on temporal, spatial and SNR scalability of SVC stream.

Human perception of different temporal resolutions has been investigated for a relatively long time. Especially, much work has been done to find the minimum acceptable frame rates of video stimuli for various tasks such as target tracking, target detection/recognition, lip reading, orientation judgment, etc. [CT07 . As for the scenario of video consumption, various factors affect the perceived quality of different frame rates, e.g. content type, viewing condition, display type, and observer characteristics. The video assessment can be performed considering the above factors in order to measure the appropriate frame rate for different content type.

For a certain target bit rate, once the spatial resolution is fixed, the temporal resolution can be increased only at the cost of decreased frame quality. The video assessment should be performed in order to examine the optimal trade-off between these two dimensions in terms of perceived quality of the final video sequence. Traditionally, it is believed that a high frame rate is more important for content with fast motion than a high frame quality, which is supported by [YMH03]. In [YMH03], subjective experiments were conducted for video sequences encoded by using three different codec's (i.e. the Sorenson codec 2.1, H.263+ and a wavelet-based codec) for eight content types. For a fixed resolution of 352 × 240 pixels, three frame rates (10, 15 and 30 Hz) were considered. Overall, a frame rate of 15 Hz was most preferred across different coding conditions. However, content-dependence was observed, i.e. for content with slow (or fast) motion, preference of a frame rate of 10 Hz (or 30 Hz) was nearly as high as 15 Hz. Similarly, a double stimulus continuous quality scale (DSCQS) experiment in [WSV03] compared H.263+ sequences coded at three different frame rates (7.5, 15 and 30 Hz) with five quantisation parameter (QP) values, for a fixed spatial resolution of 320 × 192 pixels. The results showed that, for slow motion content, subjective quality degradation due to frame rate reduction was only minor.

6. CACHING AND CDN

6.1 ENVISION Cooperative Long-Term Caching (FT)

The peers using ENVISION application can share long-term contents like VoD or other type of files. In that case, the difference between long and short-term is probably that prefetching is not necessary for the latter case.

The exchange of content between peers inside ENVISION overlay network use real network of several network services providers (i.e. autonomous system or AS). To accelerate download speed for each peer who requests the same content, ENVISION can collaborate with caching solution located in background network.

Network caching solution must first ingest the content (partially or completely). Content sharing between peers are analysed by the cache which is able to monitor ENVISION traffic. The cache stores the content: pieces of files or chunks or complete files following used protocol. For P2P protocol, pieces of content are treated as single files with a specific ID like chunks. In case of HTTP protocol, depending on cache mechanism, ingestion can be based on a divider which creates chunks (like P2P) or based on asynchronous ingestion for entire content. Once the piece, chunk or entire content is stored in the cache, next request for the same content will be served by the cache. In case of partial storage, the cache will serve the available pieces and ENVISION peer retrieve the other pieces directly from other ENVISION peers.

The distributed caches are also considered as network caching mechanisms which store the contents at different locations to provide a high hit ratio following popularity of requests and contents.

6.1.1 ENVISION Cooperative long-term Caching (FT)

The study that follows needs some hypothesis. Cache is part of ISP equipments and it is under its responsibility. Cooperative caching is in that case not only a possible algorithm inside ENVISION peers but also on specific ISP equipments able to improve signalling and data delivery between all ENVISION peers.

So, even if for ISP it's preferable to mutualise cache equipments for different services (e.g. HTTP caching, P2P caching, ENVISION caching), the first way to integrate the easiest possible ENVISION caches is to choose an explicit cache solution based on ENVISION protocol. This choice permits to avoid one of the main constraint of cache solution which is the divert function. As mentioned in D3.1, divert function is in charge to channel end-user request to the cache in order to analyse this request. This function often based on transport protocols is so directly to application level and cache needs only a connectivity visible for ENVISION peers. To ENVISION peers, the cache will appear as other peers with probably good characteristics (upload bandwidth, storage capacity).

To finish this introduction, here are the hypotheses to go further in this work:

- ENVISION Cache is considered as an ISP managed service (hardware, software, management)
- ENVISION Cache is part of ENVISION overlay network
- ENVISION Cache is in relation with the function named ISP Cache Controller
- ENVISION Cache is in charge to cache most popular contents and/or specific contents
- ENVISION Cache is able to give reporting on main caching aspects

Figure 11 presents a global view of the possible architecture of ENVISION cache.

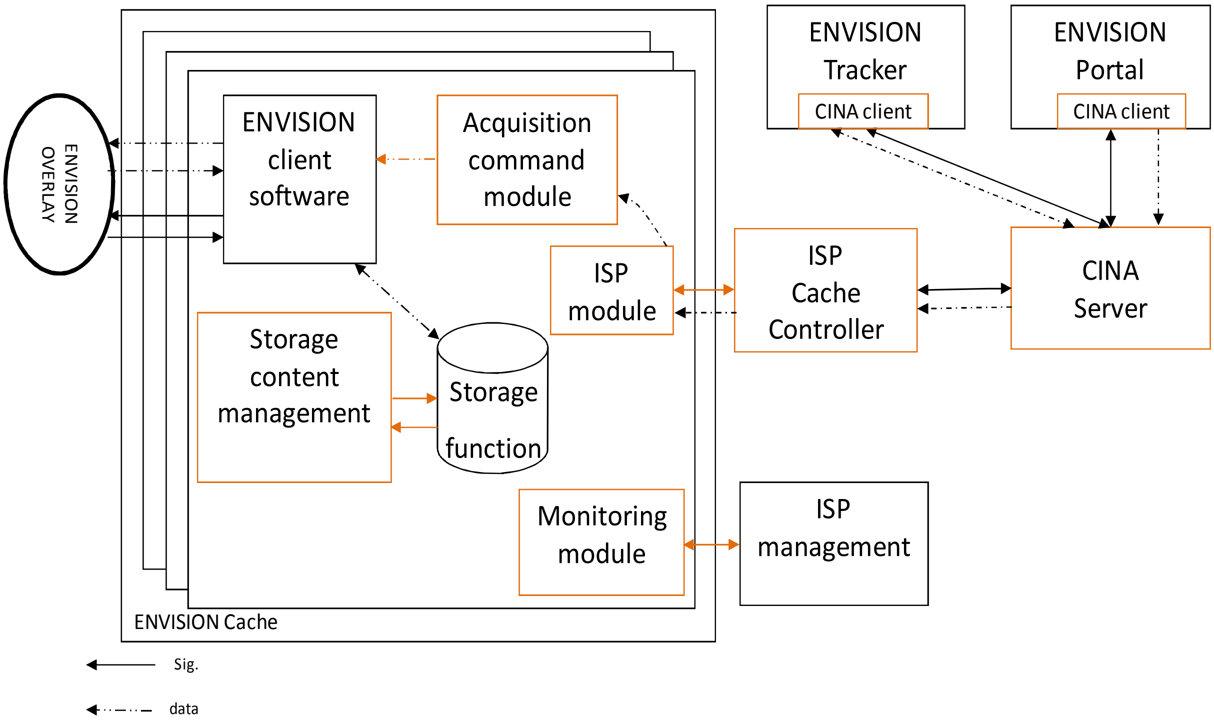

Figure 7: Global view of ENVISION cache

The ENVISION Cache integrates several modules based on ENVISION client software with added functions as Monitoring module, storage management and ISP module.

To improve the ENVISION cache ingestion, ENVISION cache will be seen in ENVISION overlay network as "super peer" or "preferential peer". For example, when an end-user gives UGC to ENVISION overlay, through CINA, ENVISION cache gets this information and starts immediately the content download.

ISP Cache Controller is in charge to get information about status of ISP caches, availability, load of CPU, load of storage. ISP Cache Controller informs ISP CINA Server the status of cache service.

The CINA interface is based on 2 elements: CINA discovery server and ISP CINA Server.

In relation with Cache service, there is an element named Cache Controller Call flows for discovery and acquisition of new content by ENVISION cache based on initial content provision by end-user source.

Figure 12 presents calls flows for discovery, acquisition and sharing with ENVISION explicit caches.

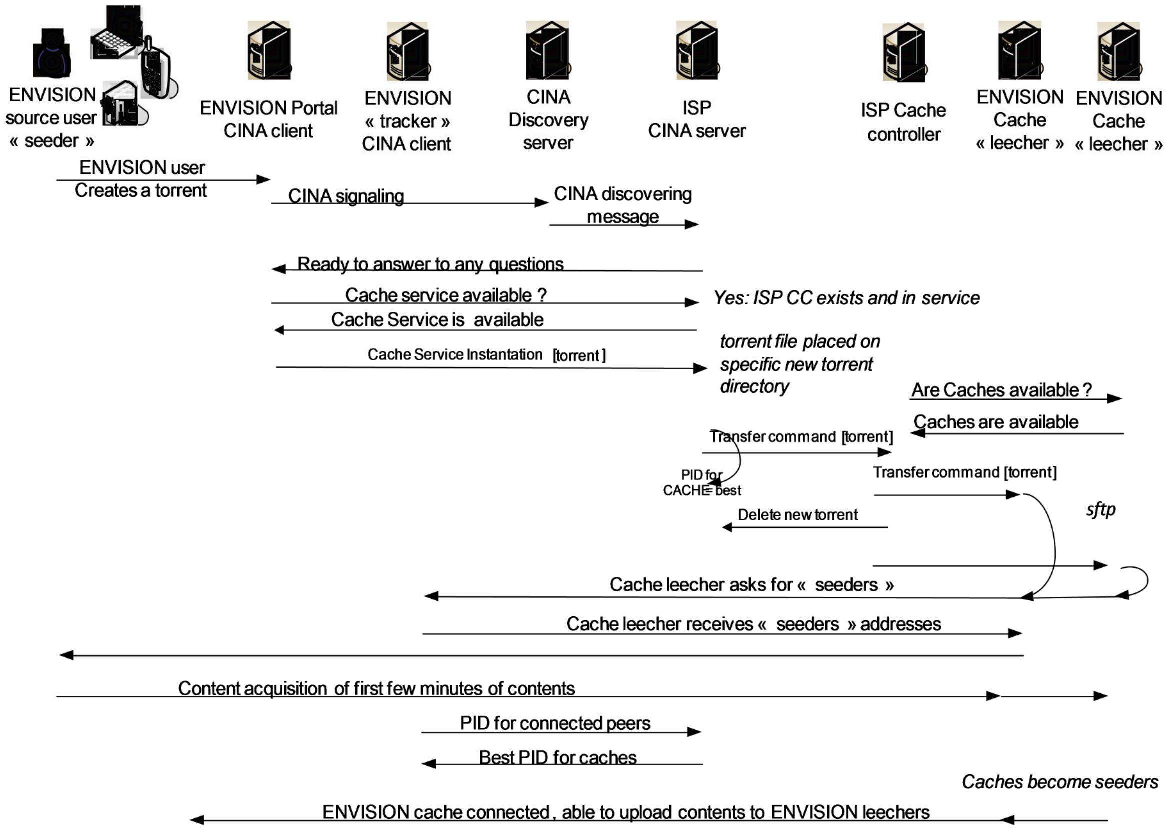

Figure 8: Example of call flows for discovery, acquisition, sharing with ENVISION explicit caches

When a new content is created by end-user, torrent is generated and available on ENVISION portal for the others end-users. ENVISION network gets information about the ISP thanks to the CINA exchanges. The ISP CINA Server sends information about the ISP Cache Service availability. Thanks to the Cache Service instantiation between CINA Server and CINA client, a new torrent can be deployed to ISP CINA Server in a specific "New torrent directory". CINA Server sends this torrent to the ISP Cache Controller. This mechanism avoids acknowledge of ISP Cache Controller address to CINA Client.

If caches are available (in service, no load on CPU, storage available, network without congestion, …), ISP Cache Controller transfers new torrent to caches. When transfer is OK, ISP Cache Controller can send a message to the ISP CINA Server to delete this torrent inside the specific directory and deletes also its proper copy.

If caches are not available, the ISP Cache Controller keeps the new torrent which will be treated as soon as cache availability is back.

Once torrent is downloaded to the cache, the different modules/peers can get content with leecher functions. When content is completely downloaded inside the cache modules, these peers become "seeders" and are able to upload the content for all ISP end-users.

Figure 13 presents calls flows for discovery, acquisition and sharing with ENVISION explicit caches with one ISP.

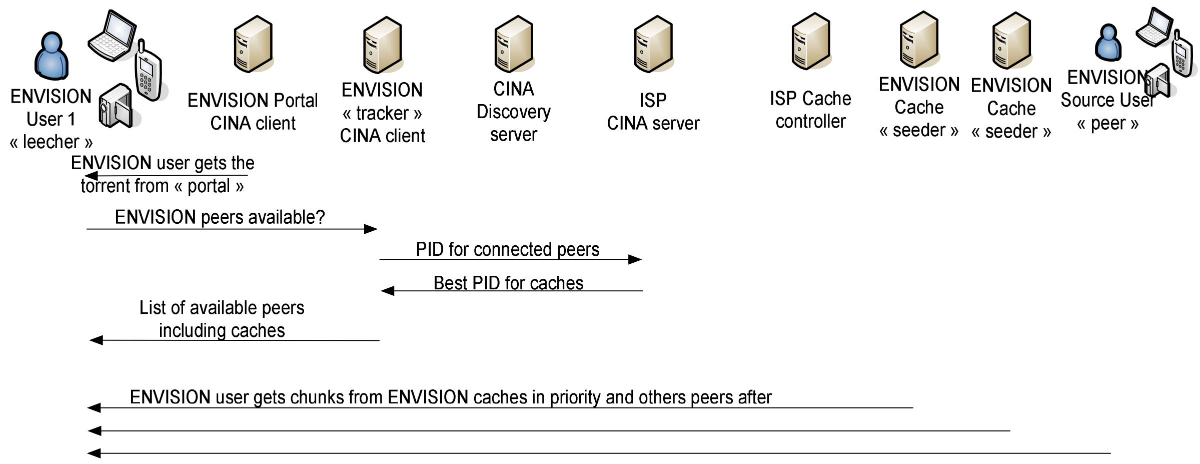

Figure 9: Example of call flows for discovery, acquisition, sharing with ENVISION explicit caches with one ISP

Caches being ENVISION peers, they are known as ENVISION peers. More, they are known with specific information as the best peers as possible thanks to ISP CINA Server. So, the ISP leechers get content from other ISP end-users but first from ISP caches.

Interfaces:

Interfaces between ENVISION client software:

- ENVISION protocols for signalling and data Interface with ENVISION ISP Interface (CINA)
- based on http exchanges Interface between ISP Cache Controller and ISP CINA Server
- based on sftp exchanges

6.1.1.1. ENVISION caches and ISP behaviour

Following ISP point of view, it is necessary to study the impact on upstream traffic generated by ENVISION caches. So, the question is: Must ENVISION cache be used to serve only the ISP end-users (i.e. as a usual cache), or to be part of the overlay network and serving any other peers (including other ISP-users)? In this case, these ISP peers may be privileged by other peers (high responsiveness, data availability …) and this will increase the ISP upstream bandwidth consumption.

In the case where ENVISION cache is used for only serving the ISP end-users, the cache will perform as a leecher for other ENVISION peers. This information must be communicated to ENVISION peers to avoid sidelining of data exchanges and cache acquisition.

6.1.1.2. ENVISION caches and cooperative algorithm

Caches as considered in chapter above (part of ENVISION overlay peers) could be integrated following two different modes:

- Simple peers with an additional module to communicate with ENVISION ISP interface module (or with http protocol)
- Identical to first one with additional cooperative algorithm

The second option permits to ENVISION caches to avoid real large storage capacities and in that way limiting the upstream traffic on ISP network.

The first option could be preferred in case of targeted QoE for ISP customers: ENVISION caches would be filled with entire contents and download will appear faster.

Depending on location and number of ENVISION caches, location of ENVISION peers in relation to ENVISION caches location, first or second option would be chosen. On the scheme illustrated in Figure 14, ISP caches are part of ENVISION overlay network, acquire the contents when sources announce a new torrent, and finally delivers contents to ISP end-users.

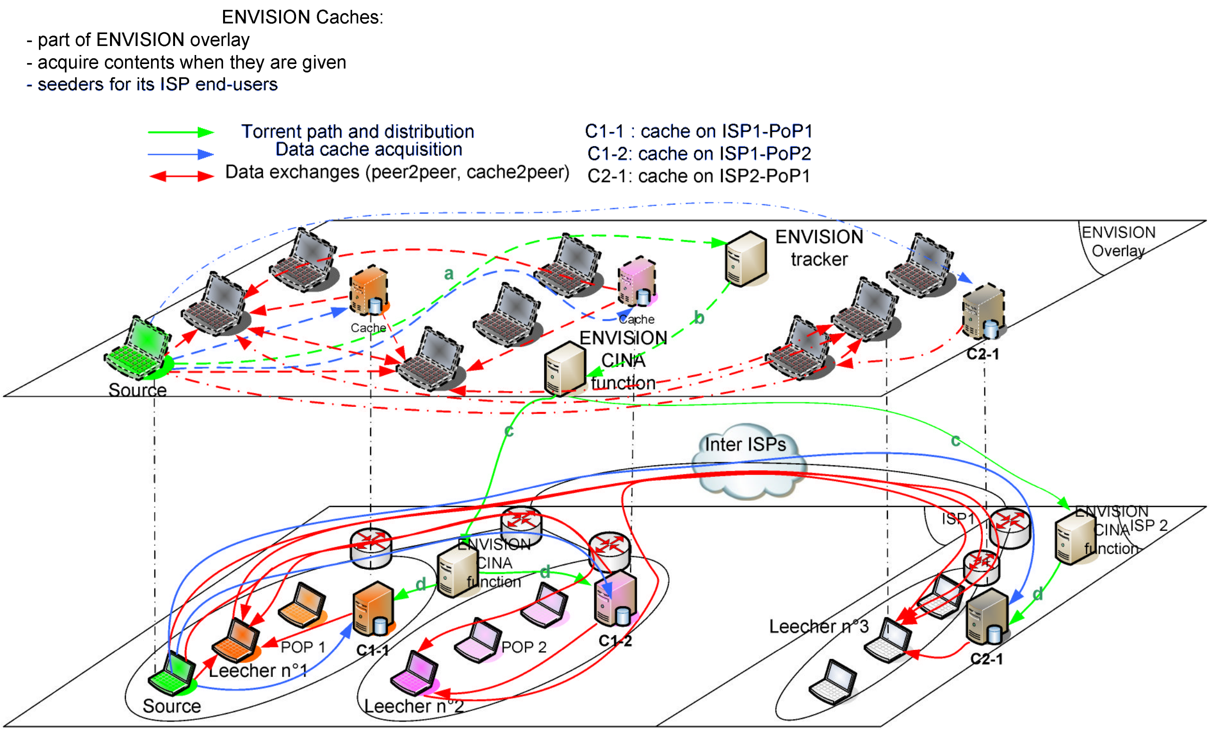

Figure 10: ENVISION peers caches integration

With specific configurations, ENVISION caches could become:

- Leechers for its local peers (i.e. internal PoP): cache behaves as real peers with or without cooperative algorithm. This configuration will improve QoE for local peers and, with cooperative algorithm, decreases the impact on local network
- Leechers for its ISP end-users (i.e. internal ISP): the QoE quality increases but the network is then more loaded
- Leechers for all ENVISION peers: caches will appear as best sources for all ENVISION overlay network and this case is probably submitted to partnership between ENVISION and all ISPs
- Seeders for all ENVISION peers: same issue as above and the result (loading links between ISPs) is contrary to one of the cache goals (bandwidth saving)

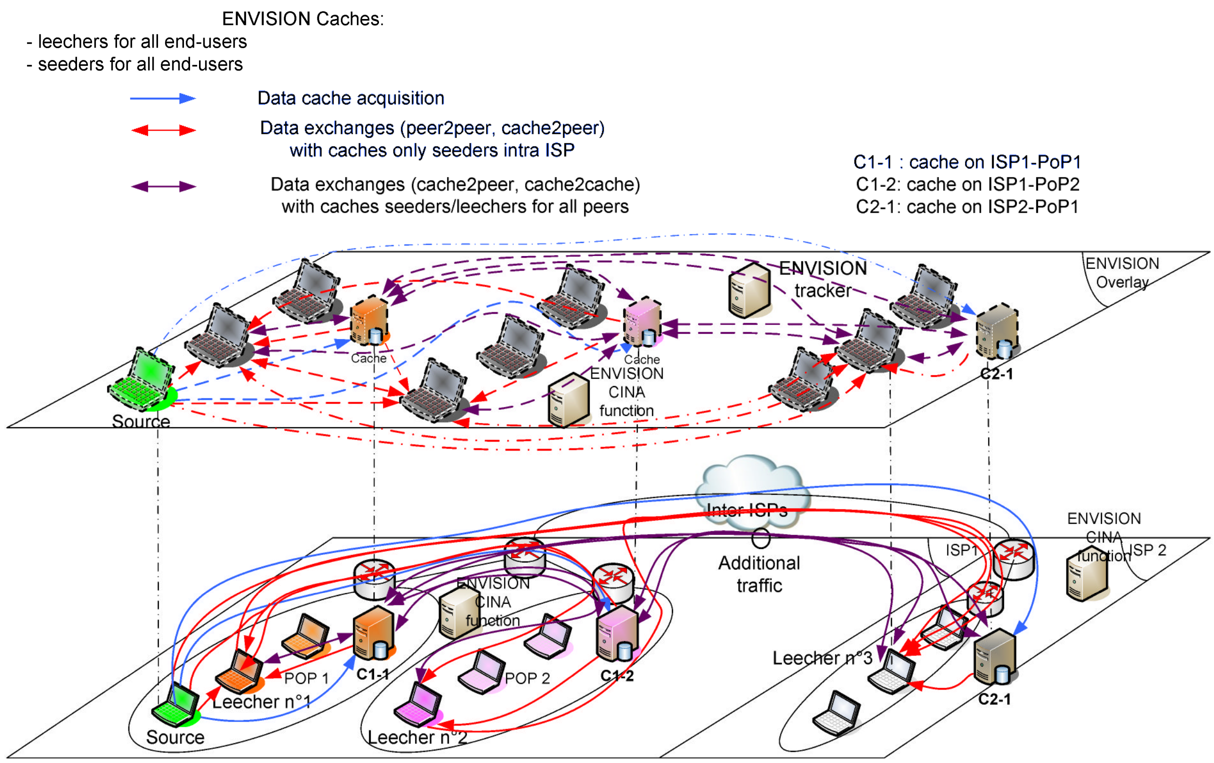

Figure 11: Traffic generated by caches based on ENVISION peers

On the scheme illustrated in Figure 15, additional traffic between for one part both ISP and other part different Pops in ISP appears when ENVISION caches are considered as entire ENVISION peers able to receive and send chunks from and to any other ENVISION peers.

To avoid this additional traffic which may introduce latency, and so causing a drop of QoE, it is possible to enhance the ENVISION tracker with a knowledge of the ISP network. Thus, the tracker will be able to locate the best peers. This solution, integrated on each ENVISION peers, can be also completed by configuration of ENVISION caches.

6.1.2. CINA-enhanced peer selection

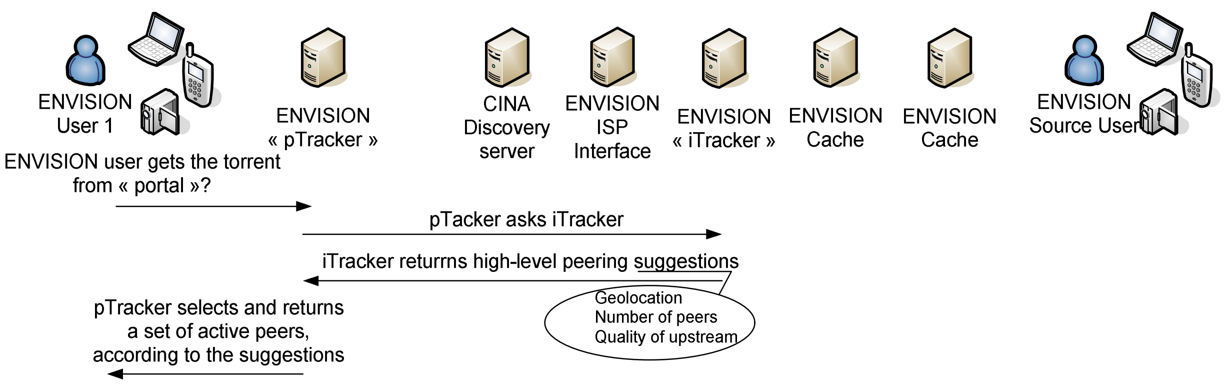

Figure 12: P4P call flows description

Figure 16 presents the call flows description:

1. Requester peer asks to ENVISION pTracker list of possible contributors for this peer.
2. ENVISION pTracker asks to ENVISION iTracker some suggestions.
3. iTracker sends to pTracker list of peers and caches.

4. pTracker returns a list of preferable peers with any specific information about caches.

The CINA interface is used in this scenario to differentiate and prioritise peers. The idea is to promote most interesting peers following the location of the source with regards to leecher. The classical tracker ("pTracker") interacts with a CINA server implementing iTracker functionality. The CINA server manages the list of connected peers inside ISP networks. The server is able to provide information to the pTracker concerning peer availability. The list in return can be based on different options as geo-location, number of peers, quality of upstream, etc. Once the pTracker gets this list, it can provide a pool of active peers according to the best options.

6.1.3. Cache Configuration Algorithm

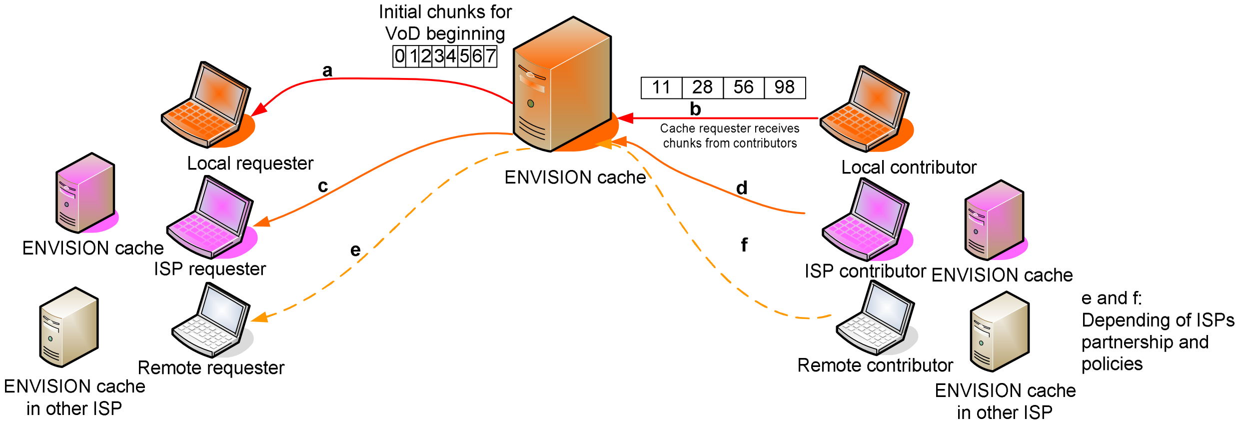

Figure 13: cache ingest possibilities

Figure 17 presents the different cache ingest possibilities. The following algorithm describes the different cases to fill a cache able to deliver contents for ISP end-users or remote end-users.

```
if {cache present and collocated to requester}
   then (!scheme a)
      if {ISP contributors}
       then
              cache configuration change from "contributor" to
       "contributor/requester" (!scheme b)
       else
              if {contributors are on other ISP}
                 then
                     cache configuration depends on ISP/ENVISION partnership and ISP
                 politics (scheme f)
           endif
    else (scheme e)
       ENVISION caches can be contributors but in opposite with P4P vision
end if
```

The algorithm insures that the cache will be filled in any configuration but with the only constraint of a previously established partnership between ISPs.

7. CLIENT SOFTWARE ARCHITECTURE

This section presents an overview of the streaming client software. The software is mainly composed of two components:

- The overlay communication component,
- The video generator/player component,

The following sub-sections present an overview of the overlay communications component (PeerNode) along two scenarios: (1) video content generation, and (2) video content consumption. Finally, sub-section 7.4 presents the simulcast support introduced in the PeerNode.

7.1 Overlay Communications

The PeerNode's logic is organised into two layers: (1) Network Layer, and (2) Application Layer. The first layer is responsible for all network connections management (Control connections and Data connections). This management is consolidated into a PeerCollection that contains all the currently connected neighbours. For each peer, some statistical information is being collected (example: RTT).

7.1.1 Main Threads

The application layer is responsible for the data distribution logic. The PeerNode is organised into the following threads:

- ***Encoder Feeder:*** This thread is responsible for retrieving the content data from the encoder and organises it into chunks to be temporarily held in the DataBuffer.

- ***Decoder Feeder:*** Respectively, this thread is responsible for extracting chunks' data from the DataBuffer and gives is to the Decoder.

- ***BufferMap Requester / Responder***: Depending on the role of a PeerNode (Source or Client), this thread is responsible for answering BufferMap Requests from other neighbours, and respectively for requesting periodically new BufferMaps from the neighbours.

- ***Data Requester:*** This thread is responsible for forming data requests given the following considerations:

    - The availability of the data in the neighbourhood,

    - The playback position, decoder position, …

    - The adaptation strategy,

    - The Scheduling algorithm

- ***Data Writer:*** This thread is responsible for handling the data response messages. It organises the received data chunks within the data buffer (discard of late data, redundancy, etc.)

- ***Data Responder:*** This thread is responsible for answering to neighbours' data requests. It also implements a receiver-side scheduling algorithm.

- ***Tracker Updater:*** this thread updates the Tracker periodically with information such as: the current neighbourhood, the liveness, packet loss, etc.

7.2 Video Content Generation

When a new content is being generated at the source, a resource description file is also generated in order to advertise the general characteristics of the video content to the consumer peers (Content Metadata: Number of Layers, Supported resolutions, Frame rates, Time/Date …). the video generator component (SVC Generator) is connected to an overlay communication component (PeerNode). The two components may be hosted either within the same program or within two separated programs.

In such a configuration, the PeerNode maintains the encoded data into its DataBuffer in sliding window time fashion. The source peer will receive from its connected neighbours either BufferMap requests or Data requests. The first request will be answered by the BufferMap Responder, while the second request will be answered by the Data Responder thread.

DataBuffer and BufferMap are jointly maintained in such a way that neighbours receiving a BufferMap response can request the advertised data chunks and get them before their delete at the source.

In parallel to the data distribution, the source peer updates the tracker periodically with information such its current live position, current neighbourhood, etc.

7.3 Video Content Consumption

To start receiving certain content, a PeerNode need to be fed with a resource description file. In this file, the peer node may find bootstrapping information (for example: Tracker address) and other information that describes the content.

7.4 Simulcast Support

In order to support simulcast transmissions, the PeerNode has been augmented with two components: (1) SimulcastServer, and (2) SimulcastClient.

7.4.1 SimulcastServer

When a multicast service is invoked/activated, the Tracker communicates to the chosen source peer the Multicaster IP address and its UDP listening port. The tracker indicates also to the source peer the content to be sent (i.e. The SVC layers to send to this multicaster). A peer, acting as a source, will instantiate a MulticasterFeeder thread and starts sending the indicated layers to the multicaster over a UDP connection.

The Peer can send different contents (i.e. different SVC layers) to different Multicasters in order to achieve the simulcast streaming.

7.4.2 SimulcastClient

When a multicast transmission is available, and if the consumer peer is allowed to receive the stream (i.e. located in the same ISP, same liveness…), can join the multicast group in order to receive the streamed content.

7.5 Conclusion

With its flexible architecture, the client software is able to handle seamlessly overlay and multicast communications in order to take benefits from network services invokes through CINA and so enhance layered video delivery.

8. CONCLUSIONS

8.1 Conclusion from this report

In this deliverable, we present the work performed in content generation and adaptation during the third year of the ENVISION project. We investigate especially a variety of multichannel algorithm for layered streaming and we provide an evaluation and comparison study between them. We also tackle the quality bottleneck problem in P2P layered streaming, either due to suboptimal neighbourhood selection and/or peer churn. We observe that the overlay formation must consider the inherent properties of the content being shared and we proposed a new overlay construction algorithm based on a preferential attachment probability for building an efficient and churn-tolerant overlay for layered video delivery.

Once the overlay is optimised based on content availability and the peers' quality level requirements, we proceed to optimise the bandwidth utilisation and reduce the useless chunks. To achieve this goal we propose a bandwidth allocation scheme based on microeconomic models namely the auction mechanism to model the competition on the bandwidth, while respecting the layers dependency. We prove the conversion of our solution to Nash equilibrium, and its effectiveness in terms of bandwidth utilisation and peers' quality level satisfaction through extensive simulation study.

Since the main goal of the adaptation process that we propose in ENVISION is to provide a good quality of experience to the end user while optimising the resources, we study in this deliverable a state of the art of the quality subjective measurement and we explore the possible assessment scenarios that can be envisaged for ENVISION.

With regard to ENVISION caching functionalities, we detail the caching architecture, giving details about its different components and their interaction with others ENVISION components.

Finally, we present the ENVISION streaming client software architecture and its two main components, namely the video generator/player component and the overlay communication component.

8.2 Overall conclusions on Metadata Management, Dynamic Content Generation and Adaptation, Adaptation and Caching Node Functions

The main goal of work package 5 is to specify and develop algorithms to adapt multimedia content delivered to a large number of end-users considering the capabilities of the underlying networks and their dynamically changing conditions. This document summarizes the final refinements of proposed techniques and algorithms.

First, we proposed a metadata model for ENVISION that describe the main actors and roles required for the content adaptation functions in ENVISION. This model organizes metadata into seven different classes (End User, Terminal, Content, Network, Service, Session and Peer). This metadata model can be seen as an enabler for any content adaptation and/or content generation techniques as it ensures the capture of all the relevant information across the entire delivery path.

Regarding the content adaptation process, we proposed novel video content adaptation techniques in order to ensure smooth content delivery as well as to guarantee an efficient content scheduling. The main goal of the smoothing mechanism is to optimize the quality of experience perceived by the end user in the presence of quality changes during the streaming session. Two types of smoothing mechanism have been proposed in the context of layered video streaming. The first one aims to reduce the layers amplitude variation (i.e. the difference of the quality change), while the second one aims to reduce the layers frequency variation (i.e. the number of quality changes within a certain period of time). In addition to simulation based performance evaluation, subjective QoE measurements have been performed. The obtained results are promising.

The proposed scheduling mechanism aims to efficiently request the appropriate layers from the proper peers. For that purpose, we modelled the problem as a Generalised Assignment Problem and we proposed a heuristic to resolve it. Then, we adapted the proposed solution to the non-layered streaming. Simulation based performance evaluation show the effectiveness of the proposed mechanisms in terms of bandwidth utilization and delivery ratio.

In order to achieve better exploitation of the available bandwidth in the network, we proposed two mechanisms. The first one is a quality level aware bandwidth allocation scheme, inspired by the auction mechanisms, while the second one is based on probabilistic approach for the construction of the overlay. The two approaches allow better exploitation of the available bandwidth in the network and provide better video quality spreading across the overlay and thus a better user satisfaction.

With regards to stored content, we discussed several existing caching techniques, both at the network and overlay layers, and we proposed a long term caching architecture and a short term cooperative caching scheme for ENVISION. The long term caching optimizes the delivery of storable content at the network level. This caching can be achieved transparently by the network operator or exposed as a service to the overlay application through the CINA interface.

For boosting the uplink capacity of a content source, we tackle the problem of using multiple links in ENVISION in order to optimize the distribution of the content over multiple wireless channels by boosting the uplink capacity of the content source. In this context, we presented novel resilient multi-channel scheduling algorithms for layered streaming to improve the video quality and reduce the packet loss and the late arrival packets.

Finally, the designed techniques have been implemented and integrated into a proof of concept software represented by a P2P layered video streaming client. The developed software has been used for the performance evaluation of other techniques related to overlay and network optimization performed within the Orange testbed.

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