# The Layer-Independent Descriptor Concept

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***Abstract*— With the fast improvements of broadband technologies, more and more demanding services can be accessed. Broadband access is supported by a range of wireless and wireline technologies. Well known representatives are WLAN and UMTS for wireless and xDSL for wireline access. However, the highly varying nature of wireless channels and the crosstalk behavior of the DSL channel are typical traits, and most challenging to combat. Moreover, the stringent requirements of real-time applications require enhancements in service delivery. The tremendous interest in communication has driven the deployment of new access systems. Hence, for the End-to-End point of view, the service delivery multifarious due to the heterogeneity of the underlying transport network. Therefore, to enable a transport independent path set-up, application and network properties that may influence the user-perceived quality need to be unified. Thereby, media scalability plays a key role. Recently, cross-layer communication is getting acceptance as a method that efficiently increases the system performance. Unlike previous works that mainly addressed the issue of application based feedback, we focus on the information that the application can provide to guide the network for local stream management. We present the necessary set of Application Layer information for scalable audiovisual streams to offer individual QoS, at lowermost packet overhead. Finally, we show the sustainable gain in system performance based on a DSL use-case.**

***Index Terms*— Cross-layer Communication, Scalable Media Encoding, Quality of Service, Local Rate Adaptation, Resource Management**

## I. INTRODUCTION

With the widespread of wireless and wireless access technologies, the paradigm of homogeneous networks, where all links are similar in terms of delay and error probability, and are substantially static in nature, does not hold anymore. We assume, the convergence of wireless and wireline networks will additionally encourage this evolution. Therefore, applications as well as the network itself, must be prepared for the emerging heterogeneous environment. The heterogeneity of networks derives from user's needs as well as from different access-network technologies. They might be based either on wireless (UMTS, WLAN, HYPERLAN) or the various Digital Subscriber Line (DSL) technologies (e.g., ADSL, VDSL). Most likely, network architectures and protocols were designed with the homogeneity in mind, following the general layered design of IP networks result in suboptimal resource and performance figures at concerned network nodes. To bridge the gap until a proper technology get deployed and be applicable to operate in such network environment i.e. DSL - Ethernet First Mile (EFM) at the link layer or Datagram Congestion Control Protocol (DCCP) [1] at the network layer, an interim as well as barrier-free solution will become important in the near and mid term time scale.

In a first step, a mediation between the state-of-the-art (media) control functionality and the various underlying network technologies need to be deployed. Otherwise the evolvement might stuck. Several methods, inspired by cross-layer design, have been proposed to improve the efficiency of transmission and enhance interaction between applications and the transport network. Major work has been done to make TCP "wireless proof", while also huge amount of efforts focused on improving audiovisual service delivery.

As mentioned before, access networks hold several problem in QoS media delivery. For wireless channels the rather fast varying channel conditions and the rapidly changing DSL channel crosstalk need to be compensated. Therefore, procedures has been developed, either to force the sender to reduce the bit-rate or use stronger error correction codes to combat these constrains. Doing that, is a highly non trivial task, due to the stringent delay

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requirements for real-time audiovisual service delivery. Even lower layer sensors are aware of these constrains, in the past resource management did not fully utilized this knowledge and considered the worst channel conditions only. Hence, the resource configuration was rather static than dynamic and resources remained partial unused during the connection lifetime.

The renaissance of hierarchical layered media encoding admit to leave the dilemma of spatial-temporal channel phenomena. Each layer of the hierarchical structure is inherently assigned with priorities, ranging from the base layer to one or more enhancement layers. The utilization of a new spectrum allocation concept introduced for DSL, allows to exploit the unused higher frequency domain. Our work addresses optimization of multimedia communication over heterogeneous networks with the focus on DSL access networks.

In this article we introduce a general cross-layer signaling scheme that is used to distribute forward information along the user plane path. Network elements that sensors network congestion or worse channel condition may use the forward information to allocate resources or adjust the media stream locally according to application and network constraints. We show the achievable gain in DSL access networks by utilization of Cross-Layer Resource Management (CL-RM) and Unequal Error Protection (UEP).

This paper is organized as follows. In Sec. III, we elaborate media delivery functions worth to be optimized. Next, we present an overview of cross-layer communication schemes and list requirements to be considered, see Sec. IV and Sec. V. In Sec. VI we present a novel framework for scalable media delivery and briefly describe the concept behind. First obtained performance gains, Sec. VII, and a practical example based on a DSL usecase will be presented, Sec. VIII. A summery concludes the paper. In the next section we pose the motivation about this research activity.

## II. MOTIVATION

With audiovisual traffic, unlike TCP-based applications (e.g., file download) which demand lossless transmission, a certain level of data loss is acceptable. Traditionally, transport networks assume equal loss importance for each data packet and all the data in a single packet is also assumed to be equally important (i.e., each bit has the same receiving probability). For multimedia traffic this assumption does not hold. In general, consecutive packets carry data of different importance for the user-perceived quality. Moreover, the contents of each packet may contain data of various importance to achieve a certain play-out quality. That is, although data loss due to congestion or bad channel conditions might be tolerated, what kind of data is lost highly matters. Cross-layer information is needed to guide, i.e. the adaptation of multimedia services, since only the application (codec) is in possession of information about the content and structure of packets while adaptation should be performed at the lower layers of remote network entities. Furthermore, congestion control and fairness among streams are serious, while solutions and deployment is essential for the network performance, thus user quality. For example the TCP slow-start process that halves the transfer window size immediately if a packet loss is detected. This built-in function of the TCP protocol is important to prevent a potential congestion collapse [2], but severely degrades the user perceived quality.

Unlike TCP, a receiver based congestion control that acknowledges the status or statistics about received packet is common in use. However, for an unreliable packet transfer mode, i.e., like UDP this is not that trivial, since the application will detect the packet loss first. Hence, an acknowledgement to the sender will be delayed, while at the same time the receiving application assume network congestion. In this context, local adaptation would blur the source of packet loss, caused by discarding packets either for the purpose of rate adaptation or resolving network congestion. Acknowledgement for unreliable packet streams as proposed by DCCP does not fetch this issue either.

To exploit the powerfulness of local packet stream manipulation and to prevent from bastardizing application layer feedback, cross layer communication is required to coordinate function located at different layers within the protocol stack, probably located at different network nodes. An extension of the signaling protocol, in particular extending the Session Description Protocol (SDP) to describe the layered encoded streams will function, but the large SDP structure would render the achieved gain. The applicability for the Offer/Answer model [3] is questionable too. Considering real-time services, we are forced to look for an efficient in-band signaling method.

From today's Internet design's point of view, it is quite serious to go for an end-to-end solution. Since this approach hides many "unsolvable" issues, particularly packet encryption, integrity, authorization etc. However, with hierarchical layered media encoding and delivery supported by cross-layer communication, concerns will deflagrate. A suitable framework will be presented in the following chapters.

## III. CROSS-LAYER APPLICATIONS

There are two approaches for source-rate manipulation; (i) codec or codec mode change (rate adaptation (RA)) and (ii) rate-scaling (RS). Encoders are usually optimized for a certain bit-rate or bit-rate range, at which the codec operates most efficiently.

If the data-rate must be reduced a new codec might be selected that operation range is within the desired bit-rate. Codec change is a costly action, as the new codec must be negotiated between the communicating endpoints. This signaling introduces delay that may result even in temporal play-out stalls. The other solution could be to use codecs that operate at multiple bit-rates (i.e., multi-rate codecs), hence only a mode change would be necessary for adaptation. A completely different approach is scaling. Scalable codecs generate highly flexible data streams. Data-rate can be simply reduced by discarding some bytes from the stream. This way, adaptation can be easily realized without introducing any signaling overhead or delay. Additional benefit of scaling is that adaptation can be carried out inside the network by low-complexity network entities that simply discard packets/layers selectively. Moreover, scaling can be used to adapt not only to persistent congestion (long to middle time frame overload or bad transmission conditions) but adaptation to temporal problems (small time frame), is also possible. Since local adaptation can be initiated without the need of the time-consuming notification of the source entity.

Due to the error-prone nature of wireless links and the heavy crosstalk phenomena in DSL loops, error protection is essential for transmission. Because of the varying conditions the required error protection varies as well. Schemes can be categorized as; (i) Equal Error Protection (EEP) or (ii) Unequal Error protection (UEP). While EEP assigns the same grade of protection, i.e., the same amount of redundancy, to each bit, UEP differentiates various protection levels. Currently, Unequal Error Protection (UEP) is being actively researched, since major performance gain is inherent to UEP. This is especially true for audiovisual services where a certain grade of data corruption and loss might be acceptable. Out of the two error protection approaches we focus on UEP, since it can achieve a better tradeoff between bandwidth consumption and transmission efficiency. However, it is likely to support EEP as well.

## IV. Overview of cross-layer communication schemes

Given the strict layered network architecture, for cross-layer communication unconventional signaling approaches must be considered. From methods only slightly loosen the layering to those that completely violating inter-layer communication constraints, a wide range of cross-layer communication schemes can be considered. While slightly relaxing the constraints might provide little to no “cross-layer” benefit, completely breaking the “rules” might result in a complicated and unmanageable “spaghetti” architecture. This latter problem is one of the fears of engineers who object cross-layer design [4]. However, between the two extremes the picture is very subtle. In the following we summarize the most interesting approaches for cross-layer signaling.

- *Interface extension between adjacent layers* For interoperability and for economic efficiency (i.e., competition among vendors) the interface between neighboring layers is well defined and standardized. However, if a boarder set of information could be signaled through the interface valuable information would be provided from higher to lower layers and vice versa. A particular example to such interface extension is the standardization work at the IETF on Dynamic Network Attachments and IP mobility, and parallel at the IEEE the 802.21 standard. Amongst others, the ongoing works focus on the definition of link layer triggers towards the network layer to aid hand-over and in general to allow adaptation to wireless coverage and quality changes.
- *Using the IP header for cross-layer signaling* The network layer is probably the best candidate to deliver and distribute cross-layer information. Moreover, as IP is thought of as the common element of future’s networks, IP is a natural choice for the basis of cross-layer signaling as well. In [5] the authors described a method that makes use of IP header options to provide forward information (called hints) to network elements and to indicate the need for feedback at certain events (called notifications).
- *Network information broker* This approach uses dedicated cross-layer information servers to collect and distribute, e.g., wireless channel states, network capabilities, and application properties. Each layer monitor its congestion states, error characteristics, actual capabilities, etc. and reports significant changes to the broker, using, for example, the Internet Control Message Protocol (ICMP). The application can also provide its descriptions and characteristics to the server. Network elements may query the broker for information whenever necessary. The main drawback of this solution is the introduced signaling overhead. Since cross-layer queries are needed especially when links become congested the generated signaling messages increase congestion, and the responses can be severely delayed. Hence, this approach is not applicable to handle transient congestion but might be used for a coarser configuration of adaptive services.
- *Local information broker* The local information broker is based on a the same idea as the network broker. However, the local broker is intended to allow for cross-layer communication on a single network entity only. That is, the broker accepts state information updates from the local layers and grants access to these data for the local layers only. Since data is maintained locally in case of congestion it can be accessed with low delay and low overhead.

It is important to note that the introduced schemes are not stand-alone solutions for cross-layer communication. That is all of them relay on certain functionality of other schemes. However, to our knowledge, how a preferred cross-layer signaling method should be synthesized has not been investigated yet. We conclude that combining IP header options and the local broker approach looks most promising to jointly address cross-layer communication between different nodes and within network elements. In the next sections we present our general cross-layer signaling framework based on these two schemes.

## V. Requirements posed against cross-layer signaling methods

The following requirements must be considered when designing a cross-layer signaling framework.

- The layered architecture should be violated as little as possible.
- Ease of introduction should be one important design guideline.
- Backward compatibility promotes and eases deployment of the new functionality, while provides means for inter-operation of network elements that are aware of, and those not supporting cross-layer communication.
- Low complexity. Network nodes and signaling mechanisms must be as simple as possible.
- Low overhead. That is, signaling, processing, and storage overheads must be reasonable.
- The achievable performance gain must be investigated. The following question must be answered. What is the benefit of introducing the new functionality compared to the price of deployment? This is generally a difficult task as the preliminary assumptions severely influence the outcome of the study. However, such results, preferably based also on business case models, would aid implementation and the wide deployment of a scheme by equipment vendors and service providers.
- Reuse existing protocols and mechanisms whenever possible. Hence, development and deployment costs and efforts could be kept at minimum.
- A solution must be general suiting today's needs as well as future requirements. From multimedia services point of view, the solution should be codec independent permitting use with arbitrary codecs available now or developed in the future.
- Preferably a proposed solution should be complete, meaning that the method could be used for communication between any arbitrary layer. That is, there should be no need for a parallel deployment of cross-layer signaling schemes.
- Considers an update of statistics and performance figures if neighboring functions would influence them.

## VI. The Layer Independent Description Framework

With the deployment of LID functionality [6] it is simple to increase the system performance, to exhaust the number parallel audiovisual streams or to fully exploit the available bandwidth resources. Media Gateways, Mixers, wireline/wireless Access Points (AP) might be upgraded first, unless network or edge routers would support the LID. The LID is inherently scalable, supporting fewer complex to sustainable high quality solutions. Enabled by the flexible grouping of "Preference Descriptors".

### *A. Concept Considerations*

For Local Adaptation (LA) forward information must be provided to permit application-driven differentiation among packets under challenging temporal conditions. A network entity do not know the importance of the packets from the application's point of view. Of course, accessing the application specific Real-time Transport Protocol (RTP) [7] headers might provide some hints about the contents of the packet but relying on this method is highly undesirable. First, it is very problematic for a network entity to reliably guess the present higher-layer headers, as well as application specific extensions. Second, with the appearance of new header formats, e.g., for new codec's, every network entity that utilizes header parsing for collecting information must be upgraded to be able to interpret the new specifications. Hence, dedicated, procedures are needed to gather information about the user-perceived quality influenced by packet loss or packet corruption, to improve the delivery of real-time services. Collecting quality information is costly in time and effort, thus it is worth to investigate procedures that introduce generality.

In the context of Local Adaptation (LA), it need to be emphasized that LA can be utilized for scalable or adaptive media streams only. Moreover, layers below the application layer need to be aware which stream is scalable, adaptive respectively. Therefore, we propose a Layer Independent Descriptor (LID) that describes a collection of parameters required to deploy RA and UEP. The description is based on a modular, media agnostic style, optimized for hierarchical layered audiovisual streams. [1] It has to be noted that the presence of forwarding information (LID) and the binding to a packet stream represent the cornerstones of this concept.

Actually, the current RTP specification suggests more appropriate methods for handling layered encodings. The basic methodology suggest using different UDP port numbers and a different session for each layer. A disadvantage that its impractical in today's "NATted" and "Firewalled" world, while for the latter the main rationale is minimization of firewall pinholes. For the sake of assumptions, it is a requirement that a layered encoded media is sent over a single stream, i.e. RTP session, rather than to be spread over parallel RTP sessions. Undoubtedly, the latter is much easier to implement in term of bit-rate adaptation, error protection, etc. Next, we present in detail the LID and the binding of layers to packet streams.

### *B. The LID Structure*

The LID consists of three field groups: (i) Traffic Class group, (ii) Packet Drop Preference group, and (iii) Error Protection Preference group, cf. Fig. 1. The Traffic Class group identifies whether the service is loss-tolerant and whether adaptive and scalable applications are located at the endpoints. The Packet Drop Preference group specifies the drop preference of the packet. As for local adaptation, discarding packets is the only means of network elements to resolve congestion this information provides the possibility of an application-controlled discarding mechanism. Fields describing packet drop preference and drop dependencies are included in this group. Moreover,

[1] It should be noted that this type of information needs to pass some security checks in order to not abuse or attack wireless networks. However, security considerations are out of the scope of this paper.

packet truncation is becoming also an option for RA and UEP with the appearance of Fine Granular Scalability (FGS) layers.

For packet truncation, additional fields are necessary to indicate the possibility of data chopping. Fields specifying the offsets of truncation points might also be needed. Offsets are used to define a base-length and if required explicitly specify the chops for adaptation. The base-length defines the smallest size of payload which is still useful for decoding. No gain in terms of scalability can be reached by truncating a media payload below the base-length, in this case the entire packet should be discarded. The base-length covers the headers and those data bytes that are mandatory for decoding.

For sophisticated error protection the fields of the Error Protection Preference group can be used. Here, the requested protection level is specified along with information about the structure of the packet in the case for packet-level UEP. The structure is specified by defining the starting points inside the packet of the data corresponding to certain protection levels. This is done by assigning an offset length to each protection level. Moreover, as audiovisual data/codecs might be bit error tolerant, an indicator field for corrupted packet delivery request is also included. To manifest the applicability of the proposed LID concept, we refer to the following example presented next.

A scalable extension of the H.264/AVC, under development by the Scalable Video Coding (SVC) project of the Joint Video Team (JVT) of the ISO/IEC Moving Pictures Experts Group (MPEG) and the ITU-T Video Coding Experts Group (VCEG).

| | | |
|---|---|---|
| Traffic Class | Scalable? | : Yes |
| | Adaptive App? | : Yes |
| | ...etc. | |
| Packet Drop Preference | Drop Preference? | : 1 |
| | Drop Dependence? | : 0 |
| | Truncatable? | : Yes |
| | Base-length | : 70 [byte] |
| | Explicit Trunc. Points? | : Yes |
| | Offset #1 | : 150 [byte] |
| | Offset #2 | : 400 [byte] |
| | ...etc. | |
| Error Protection Preference | Corrupted Delivery? | : No |
| | Packet Protection Level? | : 2 |
| | UEP within the packet? | : Yes |
| | Protection Level #1 | : 4 |
| | Offset #1 | : 70 [byte] |
| | Protection Level #2 | : 2 |
| | Offset #2 | : 370 [byte] |
| | Protection Level #3 | : 1 |
| | Offset #3 | : 470 [byte] |
| | ...etc. | |

Figure 1. The structure of the LID and example values

For a particular example we refer to RTP payload format that is under development [8] which allows to packetize one or more Network Abstraction Layer Units (NALUs). NALUs has been specified in is specified in [9] and represent basic transport entities of the H.264 video codec family. For the purposes of SVC, the NALU header will be extended. Apart from other bit settings, 8 bits are of particular interest, so-called “TDQ”. The first[2] three bits (T2, T1, T0) indicate a temporal resolution. Slices assigned to temporal resolution 0 (TR-0) correspond to the lowest temporal resolution, that is, i.e. only Intra coded frames are available. If TR-1 slices are also available the frame-rate can be increased (temporal-scalability). The second three bits (D2, D1, D0) denotes the inter-layer coding dependency hierarchy at any temporal location. In other words, it specifies the spatial resolution for a scalable stream and if the picture may be used for inter-layer prediction for coding of a picture of a next larger coding layer. For example, slices corresponding to DependencyID-0 describe the scene at a certain resolution. If an additional set of DependencyID slices are available the scene can be decoded at a higher spatial resolution. The last two bits (Q1, Q0) specify a quality level (QL). QL-0 corresponds to the lowest quality. If additional QL slices are available the quality can be increased (Signal-to-Noise-Ratio (SNR) scalability). Based on these information the fields of the Packet Drop Preference group can be filled in. For example, assigning lower drop preference to the base quality layers (TR-0,D-0,QL-0) and higher to others. Moreover, if the corresponding enhancement layer includes FGS data the truncation indication field can be set along with the specification of the base-length.

### C. The LID concept summary

The LID is a general descriptor of the scalability of packets carrying audiovisual data. That is, for each packet of the media stream, generated by any audio or video encoder, a generally structured LID descriptor can be constructed, which describes the adaptation possibilities for the transport network. This way, network elements aware of the LID information must not be aware of the actual encoder in use, only the general LID description is needed for the adaptation/scaling/error protection purposes of the media stream.

The LID might include an extensive set of information for each single packet. However, the required overhead of including a LID in every packet renders this approach to be impractical. For a narrower set of information the DiffServ architecture could be utilized by assigning the most significant descriptor sets to certain DiffServ classes. For this mapping frameworks have been proposed, for example, in [10], [11]. On the other hand, mapping the information of a LID to only a handful of classes could severely limit the descriptive power of forward information. Hence, we propose an alternative solution where the whole information of the LIDs can be utilized by network elements. Next we present the requirements against the network and communication endpoints for setting up LID states in network elements during session-setup and propose a label binding procedure to reduce the overhead of data packets.

[2] In network byte order

## VII. Performance considerations

### A. Requirements posed against the transport network

In respect to the LID-Label processing, network elements can be categorized into (i) stateless, (ii) reduced-state, and (iii) stateful entities.

Stateless elements process neither the LID-Label setup signaling messages nor the Labels of the data packets. That is, these entities do not perform any adaptation based on the cross-layer information of the LIDs. Those network elements should be stateless which are mostly located inside a core network where severe congestion is not a likely phenomenon. Hence the LID supported adaptation mechanisms cannot be efficiently utilized. Moreover, since UEP features are most beneficial for the "last mile" communication link, the majority of network elements interconnected with reliable wire-line links should be stateless.

Reduced-state elements still do not store the LID-Label database but simplified adaptation procedures can already be in use. Reduced-state entities are DiffServ aware, hence differentiation among traffic classes is possible. Moreover, as proposed in, e.g., [10], [11], Assured Forwarding can be used to realize drop preference differentiation amongst the packets of an audiovisual data flow. This way, by mapping the LID to appropriate DiffServ code-points at the boundary of a reduced-state domain simplified adaptation can be provided by network elements. Reduced-state elements implement the simplification of the Packet Drop Preference group but cannot address Error Protection Preference. This is the reason why, reduced state entities are still candidates for core networks with wire-line interconnections. Compared to the stateless counterpart reduced-state elements support the graceful degradation of audiovisual services when congestion or network failures are experienced.

Stateful network elements are completely aware of the LIDs and the Labels as well as the corresponding signaling protocol. Moreover, stateful entities maintain the LID-Label database allowing for sophisticated adaptation of the data-rates as well as error protection schemes. Representative network elements that hosts a stateful LID function would be a wireless or wireline access point or an edge device for reduced state domains and, in general, a media gateways responsible for media adaptation.

A possible network realization is depicted in Fig. 2. By setting up the network from the above discussed three types of entities a good trade-off between complexity and adaptation performance can be found. Moreover, the notion of stateless and reduced-state elements allow for compatibility with older network parts as well as for a stepwise introduction of the LID functionality.

### B. Requirements against the communicating endpoints

A central element of the forward information approach is the construction of LIDs and Labels at a network entity and the distribution of the LID-Label binding over the communication path. The LID construction can be carried out within the sending host or a dedicated ingress edge device could be responsible for this procedure. The later solution has several advantages, hence we address only this approach. The advantages include; sophisticated codec negotiation features, enhanced admission control and billing, as well as, security reasons such as authentication and integrity.

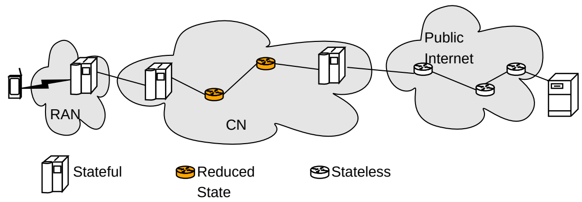


Figure 2. State requirements of network element

In regard to the LID approach the ingress has the following responsibilities. (i) After codec negotiation, based on the adaptation features of the encoder, the necessary set of LIDs must be constructed. The adaptation features of a codec can be derived using the XML descriptors specified by MPEG-7 and MPEG-21. Based on this information the fields of the LIDs can be set. (ii) Once the set of LIDs is constructed to each LID a unique Label must be associated. (iii) Then using an appropriate signaling method this LID-Label binding must be distributed to network elements involved in data delivery. We propose to use an on-path resource reservation protocol to signal the LID-Label binding. RSVP is one candidate for this purpose. For RSVP a new object needs to be defined that will carry the LID-Label associations. Network elements aware of the RSVP protocol can access the LID-Label association and install the necessary states for the flow. (iv) The ingress entity has to investigate the media-frames sent by the application, and according to the content type specified in the header of the media-frame the data is packetized and the appropriate Label is set before transmitting the packets. (v) If, because of any reason, the codec must be changed or a new media-frame type must be used during the connection, RSVP can be used to update the LID-Label states in the network. In the former case, the whole state must be substituted by a new set of LID-Label bindings, while in the later situation network elements must be notified only about a new LID-Label association. In either case, the LID-Label RSVP object can be used signaling the intention of the ingress to replace or extend the already established states of the flow.

It is important to note, that there is no restriction on the sender and receiver hosts. That is, the sender and receiver must not be aware neither of RSVP nor of the LIDs as preferably the ingress and egress entities should initiate and terminate forward information signaling. Hence, network adaptation is carried out transparently. Moreover, as discussed in section VII-A, only those network elements need to be aware of forward information signaling that can fully exploit the detailed adaptation information.

Apart from RSVP, the Next Steps In Signaling (NSIS) can be used for LID transport and Label association. NSIS is an active working group of the IETF, targeting the specification of a general signaling protocol for resource reservation and aims to overcome the scalability problem of RSVP. For the LID-Label association the QoS NSIS Signaling Layer Protocol (QoS-NSLP) need to be extended.

### C. Overhead comparison

We proposed a LID-Label binding method that increases efficiency by reducing the overhead of data packets while at the same time not reducing the adaptation information available. To underline this statement we conducted an overhead calculation comparing the naive IP header option approach [5] and the discussed LID-Label binding method. Both methods can be used to signal the LID information associated to each packet. However, the LID-Label binding procedure can exploit the fact that there are a great number of data packet but only a limited number of adaptation possibilities are allowed for a particular codec. That is, there is no need to signal the LID in each packet since most of the time the same information would be included. Moreover, by defining the LID structure and the included information in a general way allows for the applicability of the approach for any scalable codec available today or defined in the future. Hence, network elements aware of the LID do not need to be service aware but are able to perform sophisticated per packet adaptation. In Fig. 3 the LID information signaling overhead is illustrated. We assume that a single byte is used for specifying the Label, in particular, the Type of Service or the DiffServ code-point fields can be utilized for the Label association. If this approach is not possible an optional IP header might accommodate the Label. This later approach is less attractive due the increased overhead in the packets and the required processing of the optional header. In the case, when no LID-Label binding is utilized the whole LID must be included in an optional IP header, as proposed by [5]. Unfortunately, this solution would introduce a major overhead compared to our scheme. In the figure we assume 50 bytes length for a single LID descriptor. Hence, in each packet this amount of overhead is included. It can be clearly seen that as the number of transported packets increases the overhead increases severely.

With the LID-Label method the different LIDs must only be sent once, at the beginning of the session, this way the major overhead is introduced at session setup. In Fig. 3 we show four LID-Label scenarios. First, when only one LID is used, only one LID must be distributed at session setup. This would be the case with simple codecs that support only one media-frame type and scaling feature. The other examples suppose the use of multiple LID descriptors. From Fig. 3 it is clear that if a session consist of only a few packets, then due the overhead introduced at session setup, the LID-Label scheme is inferior to the IP header option scheme. However, the targeted audiovisual applications have usually a high amount of packets to transmit. For example, for video streaming at least several minutes play-time can be assumed. Even for voice telephony although the average session length is around 60 seconds, a high number of packets will be used, since generally, an encoder generates packet every 20 milliseconds from which no more than 3 are packetised into one data packet. This way, around 16 packet can be expected per second. Hence, for practical scenarios the LID-Label binding severely outperforms the header option scheme.

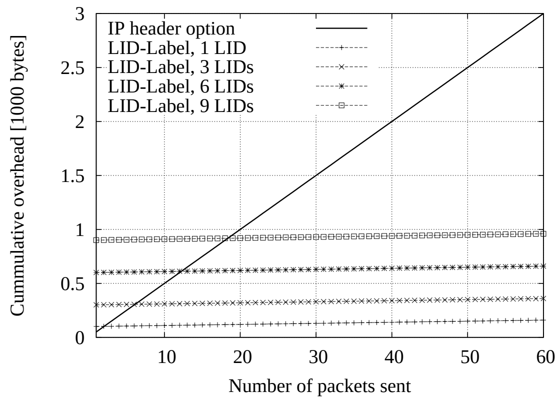


Figure 3. Protocol overhead comparison

## VIII. The DSL use-case: cross-layer information for resource allocation

### A. Current standard for DSL

The standard for DSL is developed for a static worst case traffic scenario and has only limited functions for bandwidth reservation. In the latest version of the ADSL [12] and VDSL [13] standards there are functions such as Dynamic Rate Reallocation (DRR) for negotiating maximum bandwidth and reallocating capacity between an interleaved and a non-interleaved channel but with a limited dynamic range. A new band allocation scheme, described in Sec. VIII-B below, implies a totally new DSL frame structure on the physical layer as it influences how data bits are spread out in the frequency domain. The link layer is less affected as it already today supports up to 4 different QoS channels that with small modifications can be used in a new, more advanced band allocation concept.

An ongoing work on Dynamic Spectrum Management (DSM) in ANSI T1.E1 and ITU-T SG15/Q4 will enable a more dynamic and flexible copper plant management scheme but it will still be limited by the fundamentals of the current standards.In normal scenarios gains from 10 up to 100% can be expected with simple DSM methods. However, some published papers present rather exotic scenarios where gains up to 200-300% are reported as more advanced methods are applied.

In the next subsection we present a new spectrum allocation method for time-DSM in DSL networks.

### B. The new spectrum allocation concept for DSL

In traditional ADSL and VDSL design a data stream is spread evenly over the whole frequency domain in the respective direction giving a protection against narrow band noise. Further protection is achieved through interleaving and coding enabling a transport of two channels with different QoS properties. See Fig. 4.

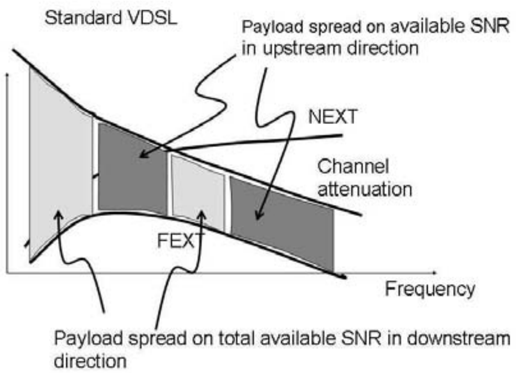


Figure 4. Payload spread evenly over the whole frequency domain in the up and downlink direction

If we look closer on the SNR distribution in the frequency domain in an operating system it is clear that SNR is higher and more stable in the lower frequency domain independent on the traffic load. The higher frequency domain offers an SNR that is lower and has a large variation over a larger frequency range. It is also highly dependent on the traffic situation/traffic load. Having this in mind it is clear that the low frequency domain is better suitable for high QoS constant bit rate channels whereas the high frequency domain is more suitable for best effort traffic and other low QoS channels.

We can take advantage of these characteristics in a bandwidth reservation (BR) concept were different data streams is allocated and spread into a specific frequency domain with a specific SNR property. See Fig. 5.

The basis for the concept is that data streams can be allocated to different parts of the spectrum giving different QoS properties. For simplification reasons we have defined two frequency regions. A lower frequency band for high quality transport and a high frequency band for available bit rate transport. Both bands can be further protected by the introduction of an interleaved channel in each band using the same interleaver depth as in the standard DSL. This adds up to 4 channels in total offering a number of channels with a much larger difference in QoS properties than the channels defined in the standard today.

The new bandwidth allocation concept together with the CL-RM framework allow for more efficient resource usage. Suppose, we have a base layer and one or more enhancement layers. With CL-RM we can specify the resource needs and QoS requirements of the layers separately. This way, at connection setup the DSL system can gather the extra information needed to map the traffic to different frequency bands. For example, a mapping scheme could be that the base layers are sent in the most reliable channel while enhancement layers are sent over less protected domains. Hence, under good channel conditions all data is likely to be received while during heavy crosstalk only the loss of the enhancement layers are to be expected. In order to identify the layer to which a particular packet belongs the DiffServ code-point or the introduced LID concept can be used. With DiffServ the mapping is straightforward, and one of the various proposals in the literature, e.g., [10], [11], is suitable to do the layer to DSCP mapping and vice versa. On the other hand, with the LID concept not only information about the corresponding layer but also detailed error protection information can be provided. Hence, with the LID, for increased complexity more sophisticated error protection can be applied to data packets that at the end can improve the transmission quality even under bad channel conditions.

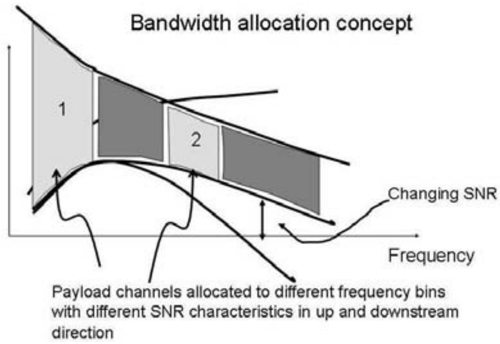


Figure 5. Payload separated into several frequency bins with different SNR characteristics

## IX. SUMMARY

Adaptation methods play a leading role in efficient service delivery over heterogeneous networks. However, Application Layer information is required to guide the adaptation. We proposed a codec independent framework for signaling such information to network elements by defining a general, layer independent data structure, the LID, that includes important transport parameters for adaptation, error-protection and resource management. These parameters are helpful to provide quality media transportation that includes bandwidth allocation, rate adaptation and protection methods. The proposed framework is useful for heterogeneous networks where time-varying parameters and unpredictable disturbing phenomena's are present such as the cross-talk in DSL or bit-errors over wireless channels (WLAN, UMTS). That is, the concept can be deployed either at wired or at wireless access networks as well. Furthermore, we examined the performance considerations of the proposed framework, and showed a special use-case scenario in DSL, where the existence of cross-talk stands strict demands towards media delivery. With the cross-layer information that the LID consists a new spectrum allocation concept can be deployed that unequally protects the different layers in the media bearing different importance.

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