On the Optimal Configuration of Metro Ethernet for Triple Play

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Abstract—Triple Play requires cost-effective implementation of three services that are different in both traffic and QoS requirements. The methods applied for optimization of Ethernet trees must support not only point-to-point pipes but also point-to-multipoint connections for multicast and statistical multiplexing models for better resource allocation.

In this paper we present a QoS architecture for Triple Play service over Metro Ethernet Networks. Based on the architecture we propose an optimization framework and a novel scalable heuristic algorithm that supports optimal off-line configuration of trees and optimal VLAN assignment to these trees. The optimized configuration considers Traffic Engineering (TE) objectives and can be performed via centralized management plane.

The obtained results show, that the performance of the proposed optimization framework can be further improved if the aggregation pipes are statistically multiplexed. Meanwhile, the proposed algorithm remains scalable, i.e. its complexity does not increase at all.

I. INTRODUCTION

In the next generation communication networks, the Triple Play service is a marketing term for the provisioning of the three services: high-speed Internet, television (Video on Demand or live broadcast) and telephone service over a single broadband connection. Triple Play is rather business model than a solution for technical issues or a standard.

The Triple Play concept assumes that integrating the services will increase opportunity for customers who may want to select between more service providers. Thus, the roles of providing network connection and certain service are differentiated. However, the same operator may act both as network and service provider.

Several network service providers are rolling out Ethernet to the home networks and fiber to the home, which support Triple Play services and bypass the disadvantages of using multiple infrastructures for the different services. This is particularly common in green field developments where the capital expenses are reduced by deploying one network to deliver all services.

A. Triple Play

Triple Play is the bundled service of Voice, Data, and Video services offered for a price that is less than the total price of the individual services. However, there is no standard for provisioning the Triple Play services, rather they are provisioned individually, since the requirements are quite different for each service. Furthermore, in our vision different services can be provided by different Service Providers, while the customers are reaching these services using the same access network.

The high speed internet access (HSI) is currently provided mostly based on PPPoE connections. The solution has the advantage of control over customer traffic, however an unnecessary level of encapsulation is introduced. Thus, another possibility is based on IPoE with Ethernet aggregation, and user traffic separation is achieved by other means like VLAN usage and/or forced forwarding to the edges. The bandwidth provisioning is asymmetric with higher download rate. Statistical multiplexing may be considered for better network resource utilization.

The VoIP is the most cost-effective solution for voice service. However, it has strict delay and jitter requirements, thus the provisioning of this service requires priority over the other services to achieve the best QoS guarantees possible. Call level admission control mechanism is also required to keep the guarantees.

The IPTV is the most promising video service for Triple Play. The basic IPTV service is the video broadcast where streams with different resolutions can be supported. The optimal provisioning of this service calls
for multicast that must be supported in the network and handled correctly. A further IPTV service is the Video on Demand (VoD). VoD does not require multicast channels; however, it desires call level CAC, since its characteristics make the VoD requirements similar to the VoIP calls. Also, the video traffic requires high level of QoS, thus besides bandwidth provisioning higher priority than the HSI must be granted for video in the network.

B. Network Scenario

The Triple Play services require end-to-end provisioning: the services should get the proper differentiation through all administrative domains. The considered network scenario consists of different domains of technologies possibly representing different administrative domains.

In the home network the home gateway distributes the services to the PC, set-top box and IP phone. The home gateway is provisioned by the service provider to differentiate the services and support three QoS channels in the first mile. This can be achieved for example with ATM PVCs in DSL.

The access nodes aggregate the traffic of several customers. Then, the traffic is further aggregated by metropolitan aggregation networks (MANs) and distributed among the edge nodes. Through these edge nodes the aggregation network is connected to the service providers. Each provider can use more edge nodes to distribute the services. In this scenario several providers are assumed and they can provide the same service presenting a competitive situation.

The role of the Metropolitan Aggregation Network is very important, it aggregates the traffic of hundreds of thousands of customers and spans from the first mile to the service edges. Further regional or core networks may be connected to the Metro aggregation, however in this paper we focus on provisioning of the Ethernet Aggregation to support Triple Play services.

II. TRIPPLE PLAY IN METROPOLITAN NETWORKS

A. Promising Network Technology: Metro Ethernet

For decades Ethernet has been the most widespread technology for LANs because of its benefits: it provides high speed communication with low establishing and operational costs. This made the Ethernet the ultimate solution for LANs. Nowadays, the transmission speed of Ethernet has reached 10 Gbps that makes it an alternative when building a metropolitan network. Since the Ethernet was born in LAN environment, it faces with several issues in connection with providing carrier grade services.

The typical access network calls for quality of service and better traffic control. A simple class-based quality of service is provided by the IEEE 802.1Q standard, which adds 8 QoS classes to the Ethernet. This standard also increases the scalability by segmenting the network into independent Virtual LANs (VLANs), each representing a different broadcast domain.

The issue of resilience is addressed by the IEEE 802.1D Spanning-Tree Protocol (STP) [1]. This protocol defines a loop-free logical forwarding topology over the meshed physical topology, that interconnects all nodes. The shape of the tree is influenced by a set of administrative weights assigned to the ports (port costs). The standard proposes a default cost set that is based on the link speeds. The main drawback of STP, that resides in its unacceptably high convergence times (up to 60 seconds), is addressed by the Rapid STP (RSTP)[2]. RSTP decreases the failing time to a few seconds. Nevertheless, both the STP and RSTP use a single spanning tree topology over the network, and forward packets of all VLANs along to this tree resulting in suboptimal path selection and inhibiting the use of redundant links.

The Multiple Spanning Tree Protocol (MSTP) [3] steps ahead introducing two major improvements: the concept of smaller administration parts or regions and the existence of several independent protocol instances (MSTIs). Inside an MST region there may be several MST instances, while the regions are interconnected using a central tree instance. In a region each MSTI runs a separate RSTP protocol instance using different port cost sets. The VLANs are uniquely associated to the MSTIs inside a region while one MSTI aggregates the traffic of more VLANs. Besides the clear advantages of supporting resilience and TE, the MSTP protocol has increased complexity and requires each MST instance to have its own link cost and priority settings. Furthermore, VLANs
must be uniquely associated to MST instances within regions. These tasks may involve a huge configuration work, which places a burden on network operators.

However, the resilience in MSTP still relies on RSTP convergence. To achieve further increase of availability protection switching based methods are needed instead of the restoration based provided by the RSTP. Such methods are presented by several papers [4][5][6].

Although the IEEE 802.1Q introduces 8 priority classes it does not define methods (capacity allocation methods, schedulers etc.) to provide the desired level of QoS. Nevertheless to support the QoS requirement of highly different services like VoIP and real time broadcast complex architectures need to be developed.

B. QoS Provisioning in Metro Ethernet

Since the Ethernet differentiates 8 traffic classes (having 3 p-bits) the DiffServ model is a logical choice for QoS. For compatibility with ITU-T and 3GPP the IP-MUSE recommends traffic classification differentiating 4 classes: Real Time (RT), Streaming, Transactional and Best Effort (BE) [7]. The three provided services have to be mapped to these four classes. The VoIP service will have the highest priority provided by RT class. As its name implies the IPTV will be mapped to the streaming class. Finally, the HSI will be transmitted as BE traffic or even Transactional traffic when QoS guarantees are required.

Since all 802.1Q compliant switches implement a priority based frame scheduler, handling of these classes is based on strict priorities, i.e., the RT class has absolute priority over the others, the streaming class has absolute priority over transactional and BE, and transactional has priority over the BE. Due to the strict priority based schedulers the traffic of higher classes has to be limited in order to prevent violating the QoS and/or starvation of lower classes. Commonly multi-level traffic shaping functions are implemented not only in the Edge Nodes, but also in the Access Nodes. The excess traffic of RT and Streaming classes will be dropped while that of the transactional class will be marked as BE.

In this paper we have considered global and static bounds based on link speeds, i.e., the amount of traffic belonging to a class must not exceed a predefined ratio of the link speed. For each traffic class a global ratio is defined by the operator considering the demanded QoS attributes, e.g., delay, jitter. However, the ratios we assumed are hypothetical ones: 10% for RT, 20% for Streaming, 30% for Transactional classes, respectively. QoS provisioning must make sure that the QoS traffic of a given class never exceeds its percentage on any link.

Note that, a service could require different QoS level for its up and downstream traffic, however, for simplicity reasons we assume that the traffic of a certain service flowing in both directions will belong to the same traffic class.

C. Network Architecture for Triple Play

The provisioning of the Triple Play services requires an architecture that provides QoS guarantees while allowing efficient network utilization supporting the requirements of all three service types. Furthermore, support for multiple network domains or service providers is also required for end-to-end QoS support. To support all these a centralized management is assumed in each domain as shown in Figure 2.

This architecture fits in the policy based management model: a policy is a set of rules controlling how to access to and set the priorities for the use of resources. The network management is performed according to these policies determined by the network operator. Policies for services are maintained at the Network Management Entity (NME). The NME has a central view of the topology information and resource usage in the network domain.

Resource control is done by pre-provisioned resources so called “TE” pipes. These resources are dedicated for the aggregated traffic of a service flowing between the service endpoints in the aggregation network. The pipe for VoIP is a point-to-point bidirectional pipe. The Video service uses an unidirectional multicast pipe, while the HSI service is a point-to-point pipe with asymmetric allocated capacity. Since there may be more than one providers for a given service, multiple pipes are assumed. These pipes are mapped to VLANs: the VLAN IDs are used for user identification while the p-bits indicate the traffic class.

Since the resource reservation on these pipes is logical, it is crucial to control the traffic assigned to them. Different mechanisms are used to achieve this: for voice, a call-level admission control is required to prevent calls exceeding the reserved capacity. For multicast video no admission control is necessary, more precisely the admission control issues are considered at the design phase. For HSI service, the user traffic must be properly policed and shaped before assigning it to the pipe.

The border nodes (access and edge nodes) are responsible for enforcing the given policies upon flowing traffic, thus, they called as Policy Enforcement Points (PEPs). This can be achieved via policing and shaping the traffic.

Besides the PEPs, an entity is needed that is able to perform CAC defining the policies and sending them
to the PEPs. This role is referred to as Policy Decision Point (PDP). It determines the QoS requirement of the required service and performs the CAC according to the QoS policy. When a service request is accepted, a pipe is selected and assigned to it and the policies are sent to the PEPs as configuration data. This can be done, e.g., using Simple Network Management Protocol (SNMP).

While the PEP function is distributed among the border nodes, the PDP can be either centralized or distributed. The most widespread concept is to focus all PDP related functions, including network configuration, CAC, etc., to the NME. This solution is proposed for instance in [8]. A further solution is to distribute some PDP functions (e.g., CAC) to the border nodes as proposed in [9]. However, in both cases the NME performs the network configuration via routing and dimensioning the TE pipes and setting up them within the network.

The concept of TE pipes makes the resource reservation easier; moreover, defining a logical overlay network of TE pipes also simplifies the CAC process. However, the pipes have to be accurately dimensioned and routed.

D. VoIP service: Call Level Multiplexing

When dimensioning the TE pipes for VoIP the main goal is to provide an acceptable bound for call blocking probability. The well-known Erlang-B formula determines the number of simultaneous calls required to serve a fixed size of population with a defined blocking probability threshold. Then, the bandwidth of a VoIP TE-pipe can be based on the number of parallel calls and on the bandwidth requirement of the assumed codec. However, to ensure the given requirements call level CAC is required, which is performed by the NME. The investigation of exact methods for dimensioning the pipes is not among our goals, furthermore, applying multiple different codecs does not influence the design task itself.

E. IPTV Service: Multicast in Metro Ethernet

To efficiently provide video broadcast it is crucial to support multicast; however, in a standard Ethernet all the multicast streams will be flooded just like broadcast traffic. This method wastes bandwidth. This flooding behavior can be prevented in two ways:

- **Manual multicast filters** based on VLAN or Ethernet multicast address: Service providers can configure Ethernet switches manually with a multicast filter to prevent a specific multicast stream from being sent out on a particular port. This results in a multicast architecture that optimizes the transport bandwidth in the Ethernet aggregation network, however the approach is operator intensive and difficult to manage.

- **Dynamic multicast forwarding**: Ethernet switches can also listen to (or “snoop”) the Internet Group Management Protocol (IGMP) JOIN messages used by receivers to query for a multicast source, on a certain port and then decide to pass that specific stream. This way, an IGMP snooping switch provides the benefit of conserving bandwidth on those segments of the network where no node has expressed interest in receiving packets addressed to the group address. This is in contrast to normal switch behavior where multicast traffic is typically forwarded on all interfaces. An other possibility is the use of the GMRP (GARP Multicast Registration Protocol) [3] for interworking with IGMP; however, only a few vendors support it.

However, counting only IGMP snooping the multicast trees are formed ad-hoc, thus, the operator has no methods for influencing these trees. Therefore, we use the VLANs to restrict the multicast domains to trees that can aggregate more multicast trees defined for the IPTV programs. Inside these trees IGMP snooping can be used but it is not necessary.

F. HSI Service: Statistical Multiplexing

The bitrate of typical internet traffic generated by home users varies significantly and therefore it does not exploit the provided bandwidth. This fluctuation of the traffic remains even after the aggregation of traffic of multiple users. Allocating the maximal bit rates of the individual flows leads to an underutilized network. To decrease the allocated capacity and to keep the QoS the effect of statistical multiplexing can be exploited.

F. Kelly [10] presented the theoretical basics of statistical multiplexing aware dimensioning considering different traffic models. Although these models are rather accurate, they are quite complex to implement.

S. Floyd in [11] proposed a simple method to calculate the effective bandwidth for aggregation of independent
traffic flows based on the Hoeffding bound. Since the Hoeffding bound is guaranteed to give an upper bound for traffic, this estimation is conservative. However, it works well only when large number of individual flows are assumed and the ratio of mean and peak rates are close to zero. Since the considered flows are aggregated the mean to peak ratio is not small enough. Therefore, this method would provide worse solutions, and it is excluded.

To obtain more accurate models, let us suppose that the individual traffic flows are independent and the aggregation has Gaussian distribution. Then, a simple, but effective model can be used presented by Guèrin et al. [12]. The effective bandwidth — the bandwidth allocated to guarantee that the overflow probability is below $\epsilon$ — is expressed as the aggregation of mean bit rates ($m_i$) plus the standard deviation $\sigma$ of the aggregation considered $\alpha$ times.

$$BW_{\text{eff}} = \sum_{i=1}^{n} m_i + \alpha \cdot \sigma.$$  

The desired overflow probability level can be achieved by adjusting the $\alpha$. For instance, for providing probabilities of $10^{-2}$ and $10^{-8}$ $\alpha$ has to be set to 2.32 and 5.61, respectively. The exact formula of calculating $\alpha$ can be found in [13]. However, to use this model not only the mean rate of the flows, but also their variances are required.

G. Configuration Task

Besides the bandwidth management the key role of the NME is to set up the network via determining routes or multicast trees of the VLANs describing the TE pipes, while the QoS constraints are kept. This optimization is performed off-line, thus, the global view of the network configuration is known.

First, the operator defines the traffic matrix for each service. A method for dimensioning the pipes is presented in [9]. Nevertheless, in this paper we assume that the average (mean) and maximal (peak) bandwidth rates are calculated in advance for each demand.

Then, based on the presented method the NME calculates the network configuration: defines the pipes considering the defined QoS constraints. For VoIP and HSI it defines individual pipes between each pair of access and edge nodes. For video streaming one multicast tree is defined for the edge nodes from which streaming service will be provided. It also defines the trees for the MSTIs and the assignment between the VLANs and the MSTIs.

Finally, the network setup is performed using SNMP. The VLAN-to-tree assignment is done by setting up the border nodes, while the trees are defined by adjusting the port costs. For each MSTI an own port cost set is defined: For the links that are used by the considered tree will get equal low costs, while the edges that are forbidden will have equal high costs. To guarantee the desired tree shapes the high cost must be at least $|L|$ times larger than the low cost, where $L$ is set of links in the network.

III. PROPOSED CONFIGURATION METHOD

In this section we describe a method for configuration of VLANs taking into account the requirements of all three services.

A. Formal Model

We consider the network given as a directed graph $G(N, L, C)$, where $N$ is the set of switches (nodes), and $L$ is the set of the links, and $C$ is the set of link capacities. Since the Ethernet ports are full-duplex, each link is modelled by two antiparallel directed edges.

The demands ($d \in D$) describing the requirements of TE pipes, are modelled as connections directed from the access to the edge regardless of the direction of the traffic. For unicast pipes one source, while for multicast pipes more sources are given. One destination is defined for both types of pipes. The size of a demand is described with two parameters: the average ($b_{\text{mean}}^d$) and the peak ($b_{\text{peak}}^d$) rates. Due to management issues direct connections between access nodes are prohibited.

For VoIP service point-to-point and symmetrical pipes are used, i.e., one source and one sink are defined and the required capacity will be allocated in both directions. The VoIP is considered being Constant Bit Rate (CBR) traffic, thus, the peak and the mean rates are the same.

For video broadcast statistical multiplexing is allowed, and it uses asymmetric multicast pipes. Therefore for these demands a multicast tree is defined and the capacity is allocated downstream only (from the edge toward the accesses). Note, that these multicast trees are restricted by VLANs, thus, they differ from the trees defined for MSTIs. An MSTI can support more multicast VLANs and simple VLANs at the same time.

For the high speed internet service unicast pipes are used and bandwidth is allocated in both directions as in case of VoIP; however, the amount of bandwidth is different. Moreover, to exploit the statistical multiplexing gain not only the peak rate but the average rate has to be given as well for both directions.

The trees in MSTP Ethernet are responsible for routing the traffic. Since direct connections between access nodes are prohibited for security and billing reasons, the traffic surely flows through the edge nodes. Thus, the
edge nodes are a rational location for roots of these trees. Moreover, the tree root placement does not have effects on the available throughput, since using the defined cost set the shape of a tree does not depend on the location of its root.

The multiplexing gain can be exploited when the pipes are dimensioned considering the statistical behavior of the traffic as presented in [9]. The pipes should be able to transmit \( n \) independent basic flows described by their mean and peak rates. The pipes are dimensioned to provide that the probability of exceeding the allocated capacity must be smaller than a predefined variable. On the links transmitting more pipes the allocated capacity is the sum of the effective bandwidths of the pipes.

However, the aggregated traffic in these pipes still fluctuates, so we can achieve further multiplexing gain if we consider not only the peak rate (\( p_i \)) of the pipe but the mean rate (\( m_i \)) as well. To use Guérin’s model, the variation of the aggregated flow has to be defined and it has to be estimated based only on the peak and mean rates of the flows. Assuming that the individual flows are independent the variance of aggregation will be: \( \tilde{\sigma}^2 = \sum \tilde{\sigma}_i^2 \).

The deterministic multiplexing, where the allocated capacity is the sum of the peak rates of the individual flows \( BW = \sum p_i \), is presented as a reference.

**C. Proposed Heuristic Algorithm**

For solving all subtasks of the predefined optimization a simple, but effective method is proposed in this section. It performs the VLAN routing and the tree covering at the same time. The topologies of the trees are combined from paths of the assigned VLANs. The algorithm itself is a generic one since the QoS constraints and the demands are given as an input. Therefore, this method can be applied not only for the presented QoS architecture but for all architectures that assume Metro Ethernet and strict priority based QoS.

The method decomposes the problem to sequential search of VLAN routes and assignment to MSTIs. However, the order of selecting the VLANs for routing affects the quality of the solutions. This fact is called demand ordering problem and it is common in case of such decomposition. To handle this problem the proposed heuristic adopts the Simulated Allocation (SAL) [15] metaheuristic. The method systematically follows a trajectory in the state space of the problem applying an iterative algorithm.

In each step the algorithm randomly selects between adding a new pipe (**allocate**) to the partial solution and removing a MSTI and disconnecting all pipes assigned to it (**disconnect**). The selection is performed with fixed probabilities \( P_{allocate} = 0.9, \ P_{disconnect} = 0.1 \). Then it performs the selected operator. When all pipes are routed the solution is stored. After a pre-defined number of steps (here: 10000 steps), the solution using the least number of trees so far will be selected.

The **allocation** operator randomly select one VLAN among the unassigned ones with uniform distribution. Then, it tries to fit the VLAN to the already defined MSTIs. If the VLAN fits to any of the MSTIs it will be assigned to that MSTI. Otherwise, a new MSTI is created and assigned to the VLAN.

A VLAN fits to an MSTI only if the path defined for the VLAN fulfills the following rules:

1) On the link along the path neither the capacity nor the QoS constraints are violated.

2) The MSTI remains tree after adding the links forming the VLAN.

Ensuring that a VLAN fits to the actual MSTI is calculated as follows. First, the edges, on which there is no enough free capacity or the QoS constraints are violated, are pruned. Furthermore, for each node in the tree the edges that are not part of the tree but started from that node are also pruned to ensure the second rule.

On the reduced graph the calculation of the VLAN is much simpler. For unicast VLANs (for VoIP and HSI) a single path is sought from the access to the
edge node. The weight assigned to the links will be 1.0 since shortest (least hop) paths are searched. Design of more sophisticated weight functions are among our future plans.

For multicast VLANs the tree is determined by calculating independent shortest paths from each accesses to the sole target one after the other. When a path is found it is temporarily added to the MSTI and then further edges are pruned to avoid forming cycles. Furthermore, the weight of these edges are adjusted from 1.0 to a value close to zero, so that the path finder algorithm will prefer the already used links. If a proper path is found from all sources all are permanently added to the MSTI. Otherwise, all temporarily added paths are removed if one of the paths fails. This step implies the VLAN-to-tree assignment.

The disconnection process selects the tree having the least assigned VLANs, and removes it. The assigned VLANs are also deleted and transferred to the set of unassigned demands.

Note that, although the algorithm randomly selects between the operators, the probability of not finding a solution converges to zero as the number of the iteration increases, since the probability of selecting removal is less than the probability of constructing. Choosing different operator-selection probabilities only leads to different convergence. Former empirical results showed that to achieve best performance the probability of allocation should be roughly 0.8–0.9.

IV. CASE STUDY

In previous sections we presented an QoS architecture for Triple Play over Metro Ethernet and proposed a scalable algorithm for determining network configuration. Our former works [6] proved that the proposed “traffic-driven” configuration method produces high throughput gain in the topologies of practical interest. In this case study we present the application of the optimization framework and focus on the enhancements of the algorithm itself.

A. Test Case

Figure 3 depicts the considered topology on which the evaluations are performed. This topology follows regional network building concepts: it contains a high-speed ring formed by four switches (#3,#4,#5,#6) interconnected by GbE links. The two edge nodes (#1 and #2) are connected to two of these four nodes using GbE channels. Two aggregation parts are connected to the core. Both have dual homing structure: each node of the lower layer is connected to two nodes of the upper layer. This structure enables high availability and better TE via load sharing.

Generating traffic demands depends on the provided service. For VoIP and internet services one demand is defined for each pair of access and edge nodes while for video broadcast one demand is defined for each edge node.

During the simulations the sizes of the demands depend on the overall throughput (OT) and the service ratio (SR) parameters. OT is responsible for the system-wide throughput, while the SR defines how to distribute the traffic between the classes. We assume that the VoIP traffic is about 10%, the video broadcast traffic is about 30% and the best effort internet is about 60% of the total traffic. These assumptions define the SR variables: $SR_{RealTime} = 0.1$, $SR_{Streaming} = 0.3$ and $SR_{BestEffort} = 0.6$. The exact size of a demand, therefore, is generated randomly with Gaussian distribution where the mean value is the product of the OT and $SR_i$, while a moderate variance is introduced.

Taking the variance of the traffic into account ratios of the mean and peak rates ($PMR = \frac{b_{peak}}{b_{mean}}$) are introduced for each type of service. For VoIP the $PMR$ is 1.0, since aggregated VoIP pipes are supposed to have CBR traffic while call level statistic multiplexing is assumed. For aggregated video broadcast a moderate variation is supposed as the peak ratio is roughly 20% larger than the mean rate ($PMR = 1.2$). Last but not least, the high speed internet service is supposed to have the largest variance: the peak rate is roughly two times larger than the mean one ($PMR = 2.0$).

B. Evaluation of Results

The main enhancement of the framework is that it considers the traffic fluctuation in the pipes. However, does it have any impact on the resource allocation or the achievable throughput? Figure 4 depicts the allocated capacities (or network load) using the selected statistical multiplexing model in the cases of various throughput levels described by OT. The deterministic multiplexing (summing the peak rates up) is presented as a reference.
The throughput levels, that cannot be served, are not drawn on the figure.

![Allocated Network Capacity vs Provided Throughput](image)

**Fig. 4.** Allocated capacity at different traffic levels

This figure depicts that, considering statistical multiplexing not only less capacity is allocated for the same demand set, but solution was found for higher OT; therefore, about 20% higher throughput can be achieved!

Although the existence of this gain is trivial, considering statistical multiplexing influences the paths calculated for VLANs: the topologies of the trees will also be different. Therefore, statistical multiplexing becomes the part of the optimization task.

### C. Notes on the Scalability of the Algorithm

The reconfiguration of the whole network is performed off-line. However, the scalability of the method remains important especially when larger networks are considered. The proposed algorithm is iterative and each step is polynomial, which results in a well-scalable algorithm. In the investigated test cases, valid solution was constructed after 100 iterations that takes less than 1 minute, and only a slight increase of achievable throughput was measured as the number of iterations is increased to 1000 or 10000. These results show that the proposed algorithm is well scalable, so it can be used to perform dynamic TE.

### V. CONCLUSIONS

In order to implement triple play services for the home customers in efficient way, the providers should consider an QoS aware service architecture. However, due to cost considerations the Ethernet becomes the most cost effective solution for regional and metropolitan networks. We presented a QoS service architecture over Metropolitan Ethernet networks. This architecture was based on the concept of traffic engineered logical channels (or pipes) and supposed a simple but effective QoS model. Keeping the simplicity of the Ethernet was a further significance of the architecture.

The main contribution of this paper was to propose an efficient algorithm for off-line configuration of metro ethernet networks for triple play based on this architecture. This method not only deals with the determination of the VLAN pipes, their mapping to the trees, and the topology of the trees, but it implemented multicast VLAN based trees to provide video broadcast service. Exploiting the gain of statistical multiplexing is a further advantage. For this purpose an easy-to-calculate model (Guérin) is assumed. The performance of the proposed method is illustrated on a simple, but real life topology.

### REFERENCES