An Evaluation of Parameterized Gradient Based Routing With QoE Monitoring for Multiple IPTV Providers

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Abstract—Future communication networks will be faced with increasing and variable traffic demand, due largely to various services introduced on the Internet. One particular service that will greatly impact resource management of future communication networks is IPTV, which aims to provide users with a multitude of multimedia services (e.g., HD and SD) for both live and on-demand streaming. The impact of this will be higher, when we consider multiple IPTV services overlaid on the same network. In this paper, we propose a resource management scheme for a network provider that supports multiple IPTV providers. The proposed solution incorporates a new distributed routing mechanism in the underlying network that incorporates QoE monitoring. Through this monitoring process, network providers are able to provide timely updates of quality of flows for each IPTV provider. Simulation work has been conducted to validate the efficiency of the proposed solution in comparison to standard approaches.

Index Terms—Distributed routing, IPTV, quality of experience.

I. INTRODUCTION

The network research community is currently pursuing new solutions to enhance the communication systems supporting the Future Internet. It is anticipated that the Internet of the future will incorporate a multitude of services, in particular high bandwidth services including multimedia content. One such multimedia service that has attracted attention in recent times, is IPTV [1] (Internet Protocol TeleVision), which is often regarded as a distribution mechanism for digital television services over a dedicated and controlled IP network. A number of solutions have been proposed for IPTV networks to co-exist and be managed with traditional IP networks, for example a proposed solution for IPTV and Next Generation Networks was presented by [2]. IPTV was developed to compete against the growth of Cable and Satellite providers, thereby providing customers with greater value through the triple play service offering that includes broadband internet access, VoIP, as well as digital TV services [3]. The video services will include both Video on Demand (VoD in High and Standard definition), as well as live streaming of traditional broadcast content. Due to its increased popularity, and the potential for high revenues, it is anticipated that numerous IPTV service providers will emerge. While the opportunity for choosing between a number of IPTV service providers will enable competitive offerings and pricing to end users, this will bring new challenges to the deployment of IPTV services on the existing Internet infrastructure. The increase in the number of IPTV service providers, augmented with other additional services will lead to higher traffic volumes and increase the variability of the traffic load. This stems from the fact that multiple service providers will be consuming resources from the same underlying infrastructure. These challenges not only impact on the underlying infrastructure in ensuring that sufficient resources are provided to the IPTV Service Providers, but also for the providers to ensure that high quality IPTV content is provided to subscribers [4].

Due to these foreseeable challenges, one research area that is currently being pursued by Future Internet researchers is new, adaptive, and robust routing mechanism. As service traffic diversifies and increases in the future, routing in communication networks will be required to be more scalable, adaptive, robust, and efficient [5]. In particular, the routing protocol should be able to support: (i) a high number of nodes in the infrastructure networks, (ii) routing processes that cope with highly dynamic traffic, (iii) the ability to improve resource usage in the networks, and (iv) ensure that the improvement in quality is provided to end users for various types of services. In this paper, we present a new resource management scheme for network providers supporting multiple IPTV service providers. The proposed solution incorporates a new routing mechanism, known as Parameterized Gradient Based Routing (PGBR) [6], that addresses requirements (i)–(iv). In particular we show how PGBR is able to support a dynamic traffic profile that may result from multiple IPTV providers sharing the Internet infrastructure. PGBR is a distributed routing technique for core and metro networks that is inspired from biological processes. The routing algorithm is based on gradient route attraction for
flows between a specific source to destination pair, where the 
gradient is calculated from local interactions between neigh-
boring nodes. In order to support (iv), the proposed resource 
management scheme also incorporates Quality of Experience 
(QoE) monitoring, where regular updates are provided to IPTV 
providers to ensure maximum quality. Evaluation through 
simulation work will show how PGBR is able to ensure a high 
degree of QoE for subscribers, irrespective of the current load 
within the networks. 

This paper is organized as follows: Section II presents the 
related work on current approaches used for QoE monitoring as 
well as routing. Section III presents the proposed solution, and 
Section IV describes the PGBR routing algorithm. Section V 
presents the simulation results, and lastly, Section VI presents 
the conclusion.

II. RELATED WORK

The related work is subdivided into two sections, which in-
clude current approaches for QoE for IPTV services and routing.

A. QoE for IPTV Services

Since IPTV will directly compete with existing digital TV 
delivery mechanisms (e.g. satellite and cable networks), IPTV 
providers must ensure that customers are guaranteed a high and 
consistent QoE. QoE is a measure that combines user’s ex-
pectation and perception and is usually represented through a 
non-technical description (e.g. a form of ranking) [7]. In order 
to assess QoE, a number of factors must be considered, which 
includes dependability and control responsiveness in order to 
maximize Audio-Visual (A/V) quality [8]. These factors are 
strongly linked to Quality of Service (QoS) parameters in the 
network and any variation in these QoS parameters caused by 
congestion or other factors, can lead to a decrease in QoE. In 
[9], Fiedler et al. developed a generic solution for the relation-
ship between QoE and QoS. The unified solution is based on 
an exponential dependency relationship between QoE and QoS, 
and has been tested for three different applications. In order to 
accurately monitor QoE, an efficient monitoring methodology is 
required. A number of methods exist to measure video quality 
using either subjective or objective metrics. Subjective tests in-
volve the rating of video quality by viewers through active par-
ticipation. However, this technique is not feasible for a deployed 
service. Objective methods involve the use of empirical values 
to provide a rating of video quality and can be integrated as 
part of a service deployment. The Video Quality Experts Group 
(VQEG) provides guidelines on undertaking both subjective and 
objective measurements [10]. One popular example of an objec-
tive method is the Peak Signal-to-Noise Ratio (PSNR), which is 
calculated based on the differences between the original, refer-
ence signal, and the received signal. It has been noted that this 
method does not mimic perceptual features and thus should not 
be taken as a measure of perceptual quality [11]. As a result, 
new objective metrics have been developed which more closely 
mimics user perception through integration of factors derived 
from analysis of the human visual system [11]. 

Through efficient monitoring approaches, service providers 
can deduce the root causes of QoE degradation, such as factors 
affected by dependability or requirements for control respon-
siveness. Control responsiveness allows QoE monitoring to en-
able corrective actions at the source in order to minimize QoE. 
In [12], Muntean et al. developed a client software to monitor 
the service quality which reports to the server, in order for the 
server to select the most appropriate level of compression for 
a given user, and optimize network utilization. This allows the 
adjustment of the bit rate of the video stream as an effort to 
ease congestion, rather than the random dropping of packets in 
the network which will have unknown impact on the perceived 
quality at the client. 

Although corrective actions can be made at the source, the 
effect in Quality of Service degradation (e.g. delay, jitter, and 
Packet Loss Ratio (PLR)), can lead to errors during content 
playback in the form of visual or audio distortion, blocking, 
loss of A/V synchronization and perhaps loss of playback due 
to buffer starvation. A method for classification of service 
quality through the examination of PLR is provided in [13]. In 
[8], further research has been carried out, detailing constraints 
on latency, jitter and PLR to ensure satisfactory levels of QoE 
for video streams encoded at differing bit rates and using 
differing encoding techniques. Jitter and latency can usually 
be constrained through buffering in the end devices leading 
to smooth playback. However, the main challenge is to keep 
packet losses to a minimal level. Another issue that affects 
the QoE in IPTV streaming is the channel switching time for 
VoD contents. Example solutions to mitigate this problem 
is insertion of additional I-frames into the video stream to 
minimize the time between selection of a new stream and the 
initialization of playback [14], or pre-loading of contents to 
minimize the delay incurred as a result of channel change [15]. 
However, a major factor that affects QoE from lengthy channel 
switching is also excessive congestion and/or packet losses in 
the underlying network.

In summary, scalable, real-time monitoring of QoE is essen-
tial for IPTV providers to ensure that guaranteed quality is deliv-
ered to their customers. Therefore, a relationship is required to 
map between IPTV providers and underlying network providers 
for monitoring QoE. At the same time, the underlying networks 
will require efficient resource management and routing mecha-
nisms to minimize packet loss, which in turn will minimize QoE 
degradation.

B. Communication Network Routing

In recent years, research in communication network routing 
has been investigated extensively. A number of research initia-
tives of the Future Internet have specified the need for more 
efficient, scalable, and adaptive routing mechanisms in order 
to support diverse services of the future. The routing mecha-
nisms of the future will need to satisfy a number of objectives, 
where examples include: maximizing resource usage of the un-
derlying network in line with traffic demand, ensure dynamic 
resource provisioning for multiple providers, and as mentioned 
in the previous section, maximizing end user’s QoE. Current 
routing techniques use IGP routing protocols such as OSPF 
[16], [17] in IP based networks. Although OSPF is a distributed 
routing algorithm, the solution requires each node to have a 
global view of the network. Therefore, in the event of changes
(e.g., traffic demand change or failures), coordination is required by all nodes to re-configure routes. This coordination can allow OSPF to reconfigure routes dynamically based on various objectives (e.g., throughput on links) [18], [19]. However, this is not suitable for dynamic traffic demands that may require frequent re-routing. For dynamic traffic this can lead to route instability and a lengthy re-routing process in the event of network failures. A number of solutions have also proposed using optimization approaches, but this requires pre-knowledge of traffic demand, and a centralized view of the topology (e.g., Genetic Algorithms solutions in [20]) [21], [22]. Applegate and Cohen [23] took a slightly different approach to determining routes, where OSPF was used with minimal knowledge of the traffic demand.

Therefore, an alternative is to propose a hop-by-hop distributed routing approach. While distributed routing ensures, scalability and robustness, there are also inherent risks (e.g., loops during route discovery [24], [25]). In [26] Gohmerac et al. investigated adaptive multipath routing for dynamic traffic engineering and proposed a distributed routing algorithm that takes load balancing into consideration. The re-routing mechanism, however, is not load sensitive and is only appropriate for managing inter-domain routing (with few links) rather than intra-domain routing. The concept of gradient based anycast routing in wireless networks has been investigated by Lenders et al. [27], and is inspired by concepts of potential fields. The mechanism is largely based on opportunistic routing and does not cater for supporting QoS for different traffic types. Bio-inspired adaptive routing has also been investigated, and most recently by Leibnitz et al. [28]. The solution is, however, based on a central processing solution that calculates the pre-defined routes and thus is not suitable for large-scale networks with numerous nodes. A hop-by-hop load-adaptive routing was proposed in [29], where traffic can be streamed along multiple paths. Each node will perform the decision on splitting the traffic based on local node load information. However, this could possibly lead to high number of packets arriving out of order at the destination, which could lead to high requirements of re-ordering at the destination node.

Therefore, as discussed, there are a number of challenges to address in routing for networks of the future. In particular to satisfy a number of objectives that includes, scalability, robustness, adaptiveness, and maximizing QoE of end users. Our objective is to propose a routing solution that addresses these challenges, in particular to support load that will be placed by multiple IPTV providers where each will have varying traffic demand.

III. PROPOSED SOLUTION

A. Objective and Proposed Solution

The proposed scenario is illustrated in Fig. 1 and assumes a number of IPTV providers that have virtual overlays on the underlying network infrastructure. IPTV architectures usually consists of VSOs (Video Serving Offices) and VHOs, (Video Head Offices) that help distribute contents to end users in residential areas. As shown in Fig. 1, as the number of IPTV providers increases, this could potentially lead to highly dynamic traffic in the underlying network. Fig. 2 illustrates the proposed resource management scheme for multiple IPTV providers. The proposed scheme consists of a Service Provider Monitoring Interface that interfaces between the IPTV providers and the network provider. As discussed in Section II-A, there are a number of different QoE measurement methodologies. In order to enable flexibility in our proposed scheme, we allow each IPTV provider to load their own QoE measurement mechanism into the Provider QoE Measurement Module. Each IPTV provider maintains a relationship with the network provider through SLA agreements (IPTV-SLA). Let the set of IPTV providers be \( IPTV = \{ IPTV_1, IPTV_2, \ldots, IPTV_j \} \), where \( j \) is the total number of IPTV providers. During each SLA
request, the IPTV provider \( k \) specifies the required bandwidth \((BW_{C,k})\) for the VoD content \((VoDC_{C,k})\), expected quality \((EQ_{C,k})\), as well as frequency of monitor reports \((t_{C,k})\) (see Table I). The module will in turn subscribe to specific QoS parameters for the specific content \((QoS_{C,k})\) from the Network Provider. Once a request is submitted to the network provider, a flow \( id \) of the new flow is recorded in a registry. During the streaming process, the network provider will record the QoS measurements, and this will be used to calculate the QoE metric. In this particular solution, the QoE is based on a Mean Opinion Score (MOS), which gives an indication of user’s perception of the content. The Network Provider will in turn, provide the calculated MOS to each IPTV provider, to allow each provider to monitor the quality of the streaming content \((MOS_{C,k})\) for specific content \( c \) of IPTV provider \( k \). This entire process is reflected in the algorithm presented in Fig. 3.

As described earlier, our proposed resource management scheme incorporates the PGBR routing algorithm to deliver the stream from each IPTV provider through the underlying network. Therefore, our proposed solution can provide a mechanism that can enable network operators and service providers to federate and provide improved service to the end customers [30].

### IV. Parameterized Gradient Based Routing

PGBR is located at the underlying network layer, as illustrated in Fig. 1. As described in the related work section, current routing approaches are not suitable for dynamic traffic that has the tendency to fluctuate or change frequently. This motivates us to pursue a new routing process at the underlying network that will form routes through self-organization of network nodes. Through the self-organization process, a gradient based route will form for each source \((s)\) and destination \((d)\) pair. The self-organization process is based on local neighborhood interaction, where each node is able to sense the load of the neighboring node and determine the most appropriate gradient value. Once the gradient values are calculated for each node, the route discovery is based on selecting the highest gradient from the source to destination. Fig. 4(a) illustrates the local node to node interaction, as well as the route formation. Through local node to node interaction, route 1 \( \rightarrow \) 4 \( \rightarrow \) 5 \( \rightarrow \) 6 (Fig. 4(a)) is initially formed for flow \( f_1 \) that is streamed at time \( t = 1 \). At time \( t = 2 \), a new flow \( f_2 \) is streamed and takes on the path 1 \( \rightarrow \) 4 \( \rightarrow \) 2 \( \rightarrow \) 3 \( \rightarrow \) 6 (Fig. 4(b)), in order to avoid the highly loaded node 5. Therefore, the route discovery process automatically adapts as the network load changes, and is able to react to changes in a timely manner, which is crucial for IPTV multimedia flows. Fig. 5 presents the route discovery from a 3 dimensional perspective and shows how the route discovery changes as new load are added to the network. The figure shows different paths taken as flow \( f_2 \) (Fig. 5(a)), flow \( f_3 \) (Fig. 5(b)), and flow \( f_3 \) (Fig. 5(c)) are added to the network, and also shows how the gradient field in the network changes as each streaming session is established.

#### A. Parameter Definition

This section will present the various parameters of the PGBR route calculation. Before routing is performed between a source and destination pair, a hop-by-hop route discovery process is initiated. This is accomplished through the use of a discovery packet \((Id_{s \rightarrow d})\) that migrates from hop to hop, selecting the link with the highest gradient value.

In order to support this process, the nodes must periodically calculate the gradient \( G_{n \rightarrow m \rightarrow d}(t) \) for the link between node \( n \) and \( m \) for destination \( d \) as follows:

\[
G_{n \rightarrow m \rightarrow d}(t) = \alpha \Phi_{m}(t) + \beta h_{n \rightarrow m}(t) + \gamma h_{m \rightarrow d}
\] (1)

### Table I: Summary of Key Notation

<table>
<thead>
<tr>
<th>Notation</th>
<th>Meaning</th>
</tr>
</thead>
<tbody>
<tr>
<td>IPTV( k )</td>
<td>IPTV provider ( k )</td>
</tr>
<tr>
<td>VoDC(_{C,k})</td>
<td>Video on Demand content ( C ) for IPTV Provider ( k )</td>
</tr>
<tr>
<td>BW(_{C,k})</td>
<td>Required bandwidth for ((VoDC_{C,k}))</td>
</tr>
<tr>
<td>EQ(_{C,k})</td>
<td>Expected quality for ((VoDC_{C,k}))</td>
</tr>
<tr>
<td>( t_{C,k})</td>
<td>Frequency of update between the MOS calculator and the IPTV Provider ( k )</td>
</tr>
<tr>
<td>QoS(_{C,k})</td>
<td>QoS parameters subscribed by IPTV Provider ( k ) for ((VoDC_{C,k}))</td>
</tr>
<tr>
<td>MOS(_{C,k})</td>
<td>Mean Opinion Score for ((VoDC_{C,k}))</td>
</tr>
<tr>
<td>( G_{n \rightarrow m \rightarrow d}(t) )</td>
<td>Gradient between node ( n ) and ( m ) for destination ( d ) at time ( t )</td>
</tr>
<tr>
<td>( \Phi_{m} )</td>
<td>Load of neighboring node ( m )</td>
</tr>
<tr>
<td>( t_{c_{n \rightarrow m}} )</td>
<td>Link load between node ( n ) and ( m )</td>
</tr>
<tr>
<td>( h_{m \rightarrow d} )</td>
<td>Normalised hop count of node ( m ) to destination ( d )</td>
</tr>
<tr>
<td>( \alpha, \beta, \gamma )</td>
<td>Weight values for ( G_{n \rightarrow m \rightarrow d} )</td>
</tr>
<tr>
<td>( P_{d}(s \rightarrow d) )</td>
<td>Discovery packet for route discovery</td>
</tr>
<tr>
<td>PLR</td>
<td>Packet Loss Ratio</td>
</tr>
</tbody>
</table>

Fig. 3. Federated IPTV Provider Resource Management algorithm.
where $\Phi_m$ represents the load of neighboring node $m$, $l_{n-m}$ represents the link load between node $n$ and $m$, and $h_{m,d}$ represents the normalized hop count of node $m$ to destination $d$. The $\alpha$, $\beta$ and $\gamma$ are weight values for each respective variable.

**Hop Count:** The hop count value $(h_{ij})$ determines how far each node is from a specific destination, and is stored in a table. The values are static, and only calculated during the formation of the topology. The process is never performed again unless the network topology is restructured (e.g. addition of new nodes). All the hop count values are normalized between 0 to 1. We have developed an automated mechanism for creating the normalized hop count values during topology formation, and a full description can be found in [6].

**Node Load:** The next component of the gradient equation (1), is the load of the neighboring node (\( \Phi \)). The load calculation is the ratio of free capacity to total capacity of outgoing links from the node, and is represented as,

$$\Phi_n(t) = \frac{\sum_{m} (l_{n-m,F} - l_{n-m,C}(t))}{\sum_{m} l_{n-m,F}}$$

Fig. 5. Chemical gradient formation for route discovery.

Each node contains $i$ number of links. The full capacity of the link between node $n$ and $m$ is represented as $l_{n-m,F}$, while the current load of the link is represent as $l_{n-m,C}(t)$. The calculation of the load is based on a threshold, where once the load changes over a certain threshold, the node will calculate its load and emit this to its neighbor. The algorithm for this process is illustrated in Fig. 6.

The gradient is calculated based on the current link value, node load and hop count of neighboring nodes. Therefore, the gradient value will dynamically change as the load of the link as well as the load of neighboring node changes.

**B. Routing Algorithm**

Our routing algorithm is illustrated in Fig. 7. Lines 1–8 (Fig. 7) describe the calculation of the gradient, which results from receiving information of node load changes from the neighboring nodes or from link load changes in its own link. Lines 9–21 (Fig. 7) describe the calculation of the gradient, which results from receiving information of node load changes from the neighboring nodes or from link load changes in its own link. Lines 9–21 (Fig. 7) describe the calculation of the gradient, which results from receiving information of node load changes from the neighboring nodes or from link load changes in its own link.

Fig. 6. Algorithm of the node load calculation.

Fig. 7. Algorithm for de-centralized routing.
Fig. 8. Loop eliminating routing.

Fig. 9. Multi-traffic PGBR routing.

this, the route discovery is abandoned (Line 18–19: Fig. 7). Line 22–23 (Fig. 7) describes the mechanism when the route is successfully discovered, where $p_d$ will return to the ingress router, and route the new stream. The backtracking process is enhanced from the original PGBR algorithm [32], which allowed loops to occur during the discovery process.

\section{Multi-Traffic PGBR}

In real deployment of IPTV contents, each content will have different QoS requirements (e.g. Standard Definition content may be different from High Definition content). Therefore, for the various different types of content stream, a different set of $\alpha$, $\beta$, and $\gamma$ should be applied to equation (1). An example is presented in Fig. 9. As shown in the figure, there are two different streams, where one is for multimedia and other for data. Each node contains a table with the gradient value of the next node with respect to the traffic type. As shown in Fig. 9, the multimedia stream, which has a higher weighting for hop count ($\gamma$) will tend to take the shorter path (which will lead to lower delay), while the data traffic will concentrate on even weighting of the link load and node load, leading to the paths on the outer-edge of the network. Therefore, this mechanism enables network providers to re-configure weightings depending on the type of contents, which in turn will lead to change in routing behavior of the traffic stream.

In this section, we have described the PGBR distributed routing algorithm. The main functionalities of the PGBR routing algorithm is the local view taken by each node in creating the gradient formation for route discovery. Before streaming is performed, a route discovery is performed from source to destination, where the route discovery process has the ability to eliminate loops through a backtracking mechanism. Once the route is discovered, streaming will be performed. The main advantage of the PGBR routing algorithm is in its fully distributed operation, and capabilities to balance the network.

At the same time, flexibility in routing behavior can be modified by only adjusting $\alpha$, $\beta$, and $\gamma$ of equation (1). These capabilities will be evaluated in the simulation section.

\section{Simulation}

This section will present our simulation of the proposed solution. Section V-A presents the performance evaluation of the PGBR routing, while Section V-B presents the QoE evaluation comparison between PGBR and Shortest Path (SP) for multiple IPTV providers.

\subsection{PGBR Routing}

This section presents the performance evaluation of the PGBR distributed routing algorithm. Table II presents the topology parameters, while Table III presents the traffic types parameters. In total, three different random topology sizes were used for the testing (20, 100, and 200), and the topologies are shown in Fig. 10. The simulation tests of the PGBR routing algorithm were compared to the SP and ANTS algorithm. The parameters used for the ANTS algorithm is presented in Table IV. Our simulation work performed in this section is extended from the work in [6].

The ANTS algorithm uses swarm intelligence to find the best route and also uses gradient based routing, where the gradients are set through pheromone trails. We used an ANTS version which modifies the pheromone table using age and delay based on the link load [33]. The parameters used for the PGBR data and multimedia flows are shown in Table V. Fig. 11 presents the performance evaluation comparison between the three algorithms. The performance metrics evaluated during the experiments include the average blocking probability, path length.
ratio, and network load balancing. The tests were performed by increasing the load on the network, where the traffic generated was based on a Poisson process. For the 20 node topology, a small variation in the blocking rate can be seen between the different algorithms, compared to the 100 and 200 node topology. This is due to the fact that PGBR has less maneuverability with small topologies, since one of the main advantages in PGBR is the ability to use both link and node load information which increases diversity in route discovery. However, we can still see that the blocking rate for the PGBR is lower compared to ANTS and SP as the number of requests rate increases. For the 100 node topology we can see a greater variation in the blocking probability, where the other solutions start blocking at approximately 40–50 requests/s, compared to the PGBR (for both multimedia and data), which started at approximately 80 requests/s.

When the network is highly loaded (at 100 requests/s), we can observe that the PGBR will have a maximum blocking probability of 0.15, while the other solutions will have a blocking rate of approximately 0.30. Fig. 11(c) presents the blocking rate of the 200 node topology, and we can see that PGBR has a much lower blocking probability at very high load (0.05).

This shows that as the topology gets larger, the PGBR is able to utilize the spare capacity of the links much more efficiently during the route discovery process. Once again, this is due to higher maneuverability in discovering routes when number of nodes and links are high. This is also reflected in the ability of PGBR to make quicker local movements based on link load changes, as well as node load changes. At 200 nodes, the ANTS algorithm performed the worst, and the reason is because most of the ants will get lost during the route discovery.
process. Therefore, for both small and large topologies, we can see that the PGBR algorithm is able to utilize the resources in the network more efficiently.

The average path length ratio for the different algorithms is presented in Fig. 11(d)–(f). This is calculated by finding the ratio between the actual length of the discovered path to the minimum hop count between the source and destination (minimum hop count is the ideal shortest path). The aim of this experiment is to observe the degree of deviation from the ideal length. We can see that as the load increases for all topology sizes, the average path length of the data flows gets larger (at 100 requests/s, the 100 node topology had 2.4 while for the 200 node topology this was 3.5). This is because the hop count value (γ) is set at 0.4, which results in most data flows concentrating on the less loaded parts of the network (e.g., outer edge of the topology), while the multimedia stream will concentrate on shorter routes in the center of the topology (γ = 0.6).

The objective of the PGBR algorithm is to allow the routes to use resource differently depending on their QoS requirements (e.g., data can have longer delay, while multimedia should have high capacity and small number of hops). In the 100 and 200 node topology case, we can see that PGBR (multimedia), SP, and ANTS algorithm loaded the network mostly in the center parts of the topology (with the shortest routes) leading to path length ratio that is quite similar Fig. 11(g)–(i) presents the network load balancing results for the three topologies. The approach used for computing the network load balancing is the ratio of the average loads in each node to the average load of the whole topology (therefore, for optimum load balancing, the value should be as close as possible to 1). As shown in the figures, the average network load balancing for PGBR improves the balance as the load increases for all topology sizes, where at the highest load (100 requests/s), we are able to achieve close to 0.87 network load balancing, compared to ANTS and SP. We can see that for SP at 100 node topology, the network load balancing was approximately 0.4, while for the 200 node topology that value dropped to 0.2. In the case of ANTS algorithm, for the 100 node topology the average network load balancing was 0.3 and for the 200 node topology, this value dropped to 0.1. This means that for both these algorithms, as the topology gets larger the concentration of routes are in the center of the network, which reflects the reason why the blocking probability increases as the load increased. On the other hand, the PGBR is able to adaptively discover new paths and balance the network in the same process leading to lower blocking probability and better network load balancing. Simulation tests have also been performed on the overhead of the PGBR routing algorithm, in particular the searching process. The results is presented in Fig. 12, where the tests were performed for different source - destination pairs (number of hops). The calculation includes calculating the sum of the number of hops in searching the paths $S_\text{PGBR}$ and the backtracking path $P_\text{PGBR}$ (please note the backtracking path is the path used for streaming, as this path eliminates all loops during discovery). The ratio of the sum to the shortest path $((S_\text{PGBR} + P_\text{PGBR})/SP)$ of that source - destination pair, will therefore give the overhead signaling of PGBR over SP. The ideal value is 2, which means that the sum of $S_\text{PGBR}$ and $P_\text{PGBR}$ is twice the hop count of SP. This value increases as the load of the network increases. However, we can see that this correlation value is relatively small at very high network load (e.g. 2.16 for average network load of 0.8). This indicates that the PGBR has a certain degree of directionality in the searching process, and does not lead to random route discovery when the network load is high.

The simulations tests performed here are different from the work presented in [6]. In the case of [6], we tested a single parameter sets for the two traffic types. However, in this paper we have applied different parameter set for the two traffic types for the same simulation scenario (but in [6] the two parameter sets were performed for different simulation scenario), and our aim is to show the impact of one traffic type over the other as we increase the load on the network. We can see that the blocking probability of data traffic is higher than multimedia traffic, but this gap is reduced as we increase the topology size. This is also reflected in the path length ratio, where we can see longer paths for the data traffic of PGBR compared to the multimedia traffic. Therefore, this shows that tuning the parameters can allow the different traffic types to automatically use different parts of the networks but at the same time balancing the network in the process.

B. Shortest Path vs. PGBR for IPTV Service Distribution

This section presents the performance evaluation of our Federated IPTV provider over a single network provider network described in Section III. Based on our algorithm in Fig. 3, the number of IPTV providers $j$ is two. Our scenario for this case study is a number of residential areas that contains customers subscribing to a service from one of the two IPTV providers. The purpose of the simulation was to carry out a comparison of the QoE between SP and PGBR when introducing additional flows to a network where the QoE of existing customers must be maintained, through avoidance of large scale packet loss. This process represents the MOS calculator of the resource management scheme in Fig. 2. The MOS calculator uses the QoE measurement that is monitored for each flow.

The network topology used in the simulation is shown in Fig. 13, indicating the location of the IPTV providers and residential areas within the network. As the simulation progresses
and additional streams are introduced to the network, the majority of additional streams were those from the IPTV provider 2 to residential area 4. The purpose of this was to introduce additional traffic in a particular region of the network. As a consequence, certain links within the network will begin to approach their capacity and packet loss will occur. As additional streams are introduced, PGBR will be able to accommodate more of these additional streams than SP by using PGBR’s route discovery process, while maintaining the QoE of the existing subscribers. This allows IPTV providers to increase revenues by allowing additional subscribers to access to their offered services, while ensuring that existing subscribers do not suffer any degradation in their service, thus affecting their QoE.

Two forms of video traffic were used for the simulation corresponding to two different video bit rates, 5 Mbps and 4 Mbps. Initially, the traffic load on the network was established using the 5 Mbps flows and, as the simulation progressed, additional video flows were added in the form of the 4 Mbps flows to emulate a higher level of compression for these flows. Video flows were modeled as constant bit rate flows. The flows are modeled using the MPEG Transport Stream (TS) format with an application layer payload of 1,316 bytes per IP packet, corresponding to 7 TS packets each containing 188 bytes as is typically found in IP packets over Ethernet with a Maximum Transmission Unit of 1500 bytes. Links in the network were modeled as having a fixed bandwidth of 100 Mbps, with the exception of the links directly from the IPTV provider into the network, which were given 1 Gbps links so they could adequately support as many video flows as required by the simulation. These link rates coincide with the likely existence of Fast Ethernet and Gigabit Ethernet links in a real network. We believe that the results are scalable to larger bandwidths using either leased or dedicated links. Further work will be required to validate the approach in the face of contention with TCP based traffic to assess its “TCP friendliness” in an “over the top” deployment scenario.

As discussed above, the flows for the initial set of subscribers were established in the network at the outset of the simulation. As the simulation progressed, additional flows were added in a round-robin fashion to the residential areas. As discussed previously, in order to ensure that parts of the network were congested more than others, the following loading pattern was adopted. For each new set of flows originating from IPTV provider 2 (corresponding to new sets of subscribers), two of these flows would be delivered to residential area 4, two additional flows were also added alternating between residential areas 1 & 2 and 3 & 4.

Throughout the simulation, for each IPTV provider/residential area pair, network performance metrics for the video flows were monitored. In each residential area, two subscriber flows were monitored, one for IPTV Provider 1 (SP1) and one for SP2. The network metrics monitored were jitter, end to end delay, and PLR. From related work [8] it is known that once jitter and end to end delay were kept within a fixed range, as dictated by the size of the playback buffer, the effect of quality degradation was minimal. As a result, the primary factor affecting the video quality, and therefore QoE, is packet loss [34].

As the simulation progressed and packet losses began to occur, it was found that PLR experienced by subscribers were enough to disrupt the service to an extent where video playback became impossible to maintain. The histogram in Fig. 14 shows how the total number of IPTV subscribers in the network is increased during the simulation. The corresponding PLRs, calculated on a per second basis and video quality measurements are presented for monitored subscribers. Note that not all monitored subscribers experienced packet loss, due to the network topology and asymmetric load profile. As a consequence, results are only presented for monitored subscribers who experienced losses. The simulation was conducted using the Qualnet Network Simulator developed by Scalable Network Technologies [35].

In order to measure video quality based on our Multiple IPTV Resource Management scheme (Fig. 2), a simplified version of the video quality metric presented in [36] is used for the Service Provider Monitoring Interface. Our evaluation is concerned with the analysis of simply the number of seconds where a Packet Loss Event (PLE) and thus video quality degradation occurred, within a given time. The reason for this simplification is two-fold:

- Once an error has occurred, the system will require some time to recover so that monitoring on a sub-second basis is not required.
- By simply flagging each second as containing an error or not simplifies the calculation so that a real time monitor could be embedded in a network node carrying a large number of streams and still be able to make a quality assessment for each stream.
For the purpose of our simulation the specified timeframe was 10 seconds as in [36]. The equation for the calculation of QoE at time \( n \) on a scale between 1 and 5 is presented below:

\[
Q(n) = 1 + 4e^{-\frac{n}{3.5}}
\]

where \( PLE \) represents a Boolean value indicating whether a packet loss event occurred for a given second. As packet loss events occur in succession, video quality and thus QoE decreases rapidly. The value of 3.5 used in the formula is used to represent the expected quality (\( EQ_{C,k} \) from algorithm in Fig. 3) video stream as our scenario is not able to discern whether content is fast moving action content or footage with low spatio-temporal variation. Applying a weighting of 4 to the exponential, means that any encoder related imperfections are disregarded, so the MOS score relates purely to the network impairment. The histogram in Fig. 14 shows the piecewise linear increase in offered load measured as the number of subscribers admitted above the baseline of 40 subscribers. As can be seen in Fig. 15(a)–(h), as the number of subscribers increases, certain links within the network become congested leading to losses of video content being delivered to subscribers.

We can see that using PGBR and it’s route discovery process, the IPTV providers are able to add more subscribers to their services before PLE occurs. This benefits IPTV providers, while
ensuring the QoE of existing subscribers is unaffected. This allows IPTV providers to maximize revenues through increased subscriptions without the need to lease additional bandwidth on existing links. Table VI summarizes the additional number of subscribers that could be added using PGBR before PLE began to occur.

### VI. CONCLUSION

As the Internet of the future moves towards service oriented environments, new challenges are emerging in efficiently managing communication systems. IPTV, in particular, is emerging as a popular service with its main focus of delivering various types of multimedia content (VoD and live streaming) to end users. As the IPTV market becomes increasingly competitive with multiple service providers operating over a common network infrastructure, new resource management challenges will emerge for the underlying network providers. This paper proposes a new resource management scheme that allows multiple IPTV providers to interact with the network provider, while ensuring that quality is maintained for end users. The scheme incorporates a new gradient based routing mechanism, PGBR, to deliver IPTV content over an IP network. A QoE monitoring mechanism is also incorporated into the solution to allow network providers to periodically update the IPTV providers to ensure that quality is maintained. The proposed solution has been validated through simulations, showing that the proposed solution outperforms other approaches, and in the cases reported here, can accommodate 17% more IPTV customers in the congested parts of the network when compared to shortest path routing.

### REFERENCES


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