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QoS-Aware IPTV Routing Algorithms

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Abstract

The aim of this paper is to describe how QoS (Quality of Service) metrics such as packet delay can be used to optimise the routing algorithms used in a network where IP Television (IPTV) content is being distributed. We outline the usage of metric instrumentation in a network to gauge the bandwidth limits of the network and how to use this information to generate a model of network link utilisation. Furthermore, we show that as the link utilisation rates change in our network model, we can modify the network routing algorithms to optimise the distribution of IPTV content to end-users.

Keywords: IPTV, Network Management, Quality of Service.

1 Introduction

Internet Protocol Television (IPTV) is a distribution mechanism for a digital television service using existing IP-based technologies and broadband network infrastructures. Due to the fact that IPTV uses the same network infrastructures as broadband Internet and VoIP services, it is often found offered as part of a triple-play service by Internet Service Providers [1]. IPTV is designed to compete with existing television distribution mechanisms, such as Cable and Satellite Television. As a result of this competition and in order to ensure adoption of IPTV, it must at the very least ensure a similar standard of audio-visual quality, ease of use, and content availability as these existing distribution mechanisms. This constraint can be difficult to fulfil due to the shared nature of the network infrastructure, which is often used concurrently by other network applications. An IPTV service requires that the network infrastructure meets the minimum required bandwidth for video distribution, as well as additional bandwidth to allow concurrent utilisation of triple-play and/or other services.

In order to ensure an always-on and reliable IPTV service, over-provisioning of resources and/or redundancies may be designed into the network at the planning phase. This may be replaced or supplemented by performance monitoring during the deployment phase. Some methods involve audio-visual signal monitoring methods to analyse the received content and use this as an indicator of quality of service (QoS) [2]. Network QoS metrics are used to gather information regarding the current status of the network. Using data gathered from these metrics, decisions or actions can be taken to modify network parameters or network configuration to ensure degradation of services can be avoided or kept to an acceptable level. IPTV is especially sensitive to variations in network conditions due to the real-time nature of the content being delivered. QoS metrics such as data loss and transmission delay are some of the factors which are currently used to determine how well the network is delivering content to the viewer and, potentially, how the user experiences the service.

In this paper, we describe a number of existing QoS metrics and discuss how variations within these metrics affect the perceived quality of an IPTV service from an end-user perspective. The metrics that will be discussed are Delay, Jitter, and Packet Loss. We then outline an experiment involving one of these QoS metrics (Delay) and how monitoring this metric allows us to build a simple model of our network. We then vary the traffic flows within the network and show how, using our model, we can choose the optimal routing algorithm for this new network configuration.

The rest of this paper is organised as follows. Section 2 describes some QoS metrics that can be applied to IPTV service provision and how variations within these metrics affect the service. Section 3 presents our experimental setup and the network configuration used in our simulation. Section 4 presents the results of our experiments. Finally, Section 5 details our conclusions based on these experimental results, and an outline of future work.

2 QoS Metrics for IPTV

One of the major QoS metrics affecting any network service (IPTV or otherwise) is delay. This is the amount of time taken for data to travel from its transmission source to its final destination. The total delay experienced by end-users is a combination of delay caused by the processes involved in the transmission of the data. These are:

- Time taken for a signal to propagate along the physical medium
- Delay incurred while encoding/decoding the data packet
- Delays encountered en route, such as the time spent in input queues of intermediate nodes: this can be especially significant if there is a single point through which a large percentage of the network's traffic flows.
- Factors such as contention for transmission media, equipment failure, and poor routing choices can cause large increases in delay.

In terms of IPTV service provision, excessive delays can degrade end-user experience in a number of ways. If delays across the networks are higher than usual, but remain constant, users may be forced to wait a greater amount of time before their IPTV stream is available while the audio-visual content is buffered. If delays across the network are higher than usual, but continue to increase, users may eventually experience a dropoff of their IPTV service as there is not enough data filling the playback buffer to sustain a constant audio-visual stream. Excessive delays may manifest themselves also in terms of noticeably increased delays in channel changes or the delay in reception of control information required to manage the IPTV service. In IPTV systems, when a user wishes to change channel, the current stream must be dropped and the initialisation of a new stream carrying the new requested content must take place. This is in contrast the existing Cable television distribution systems where all channels are delivered at the same time. Methods have been developed for IPTV to minimise excessive channel change delay times, as well as varying the bit-rate of the stream, when faced with varying network conditions to minimise impact of end-user experience [3][4].

Another metric that is sometimes associated with large delays is the total number of packets lost within the network. Packet loss can happen for a number of reasons: one of these is network congestion. If packets are being transmitted across the network at a rate greater than the rate at which they are being received at their destinations, the total number of packets currently travelling across the network will increase. This primarily manifests itself in increased queue sizes at intermediate nodes on the network. These intermediate nodes have finite queues in which to buffer data packets before forwarding them on their next hop. If remedial action is not taken, these queues can overflow and (depending on the queuing algorithm used) this can lead to packets being dropped from the network. Retransmission of these dropped packets is sometimes not an option: firstly this places additional load on a network which is already under excessive loading conditions; secondly, due to the nature of IPTV services and the degree of congestion of the network, by the time the re-transmitted packet reaches its intended destination, the time by which the data was required may have already passed. For an IPTV service, this can lead to interruption of the services in terms of perceptible audio-visual losses, or delay in accessing other channels available via the service.

In addition to using delay and packet loss as an indicator of network congestion, we can also monitor the variation in packet arrival times at their destination. Packet Delay Variation or ”Jitter” is based on the difference in the end to end delay experienced by packets travelling across the network to the selected destination.

In a network application where packets are transmitted from the source with a fixed inter-packet interval time, if these packets experience no variation in network conditions, they should arrive at their destination with the same inter-packet interval as when transmitted. If a packet experiences a delay due to variation in the network conditions, this will cause the packet to arrive before or after its expected arrival time, as derived from the initial transmission schedule.

Due to the real-time nature of the video stream in an IPTV broadcast, any variations in the packet arrival times can cause a large number of errors in the presentation of the video content to the user. If packets arrive too late, either due to the packets being held in a queue or lost, this is known as dispersion or positive jitter. In this case, the video stream is missing some required data and may be unable to fully reproduce the original content in the required time-frame. The opposite case occurs when a packet arrives too early. This can be caused when packets are queued up and then dispatched in quick succession at some intermediate network element. This is known as clumping or negative jitter. In this case, the packets must be buffered at the receiver until they are required: the number of packets stored at the receiver is dependent on the buffer size of the receiver and steps must be taken to avoid buffer overflow.

Previous work has shown that delay and jitter can be used as indicators of congestion-based quality degradation in video distribution mechanisms, such as using these metrics to facilitate handover of video streams in wireless systems [5]. These metrics have also proved useful in a quality adaptation system for a multimedia distribution system in wired networks [6].

Once decisions are made as to where to monitor these metrics, a simple system as shown in Figure 1 could be implemented. As network parameters change with varying load conditions, metric data will be collected and combined with metric data from other parts of the network. This will then be aggregated according to some pre-defined computational process, which will present the data in a standard format to the network management component. This management component will analyse this data, make the necessary required changes (if any) and feed these back to the network.

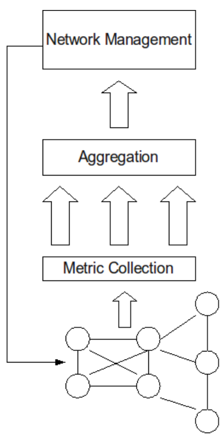

Figure 1: Network parameter modification using QoS metric aggregation.

3 Experimental Setup

The experiment was carried out using the Qualnet Network Simulator, developed by Scalable Network Technologies [7]. For our experiment, we created a network as shown in *Figure 2*. There are 2 IPTV service providers, depicted on the left side of the figure, using a shared core network infrastructure, in the centre of the figure. This core network is connected to Digital Subscriber Line Access Multiplexers (DSLAMs) which then supply the IPTV content to users in 3 different residential areas. In each residential area, there are up to 10 end-users who have subscribed to an IPTV service from an IPTV Service provider. In order to simulate an IPTV stream of MPEG-4 video traffic at a bit-rate of 4Mbps, the video stream was modelled as Constant Bit Rate (CBR) traffic, with a packet size of 1,250 bytes transmitted at an interval of 2.5ms.

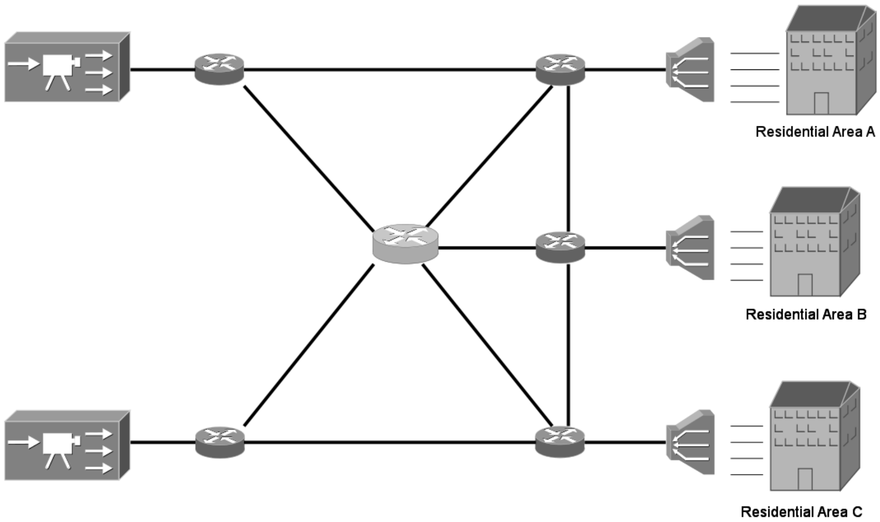

Figure 2: Network architecture used for experiment.

All network links are modelled as wired, although we can easily modify our network to include wireless links between the core network and residential areas. The links between the IPTV content distribution centres are modelled as 1Gbps Fibre links. On the wired links among nodes in the core network, the available bandwidth is up to 50Mbps, with the exception of the links between the two leftmost core network nodes and the central node (depicted in light grey), which have a bandwidth of up to 100Mbps. This is to allow a single IPTV service provider to adequately service 2 residential areas via the central node on the core network. The individual links between the DSLAMs and the IPTV subscribers in the residential units are simulated as 8Mbps Asymmetric Digital Subscriber Line (ADSL) links. No background traffic was generated as part of the simulation, as detailed below. Future work, extending the simulation to include background traffic will we carried out using an appropriate background traffic model.

During the initial design of the simulation it was found that delay would remain at a constant level, provided the link utilisation remained below about 92%. If link utilisation increased beyond this value and the network became congested, large increases in delay were seen, along with increases in packet loss.

At the beginning of the experiment, there are 10 IPTV subscribers in Residental Area A shown in *Figure 2* receiving their service from IPTV Service Provider 1 (SP1), along with an additional 10 subscribers receiving their service from IPTV Service Provider 2 (SP2). All subscribers in Resident Area C are receiving service from SP2. In Residential Area B, half of the 10 subscribers receive service from SP1, while the other half receive service from SP2. Initially, traffic from SP1 to Residental Area A and from SP2 to Residental Area C travel across the outer links without entering the central links in the core network in an effort to ease congestion.

As the experiment progresses, the number of users in Residental Area A subscribing to service from SP2 is decreased. This change in the traffic pattern is then reported via the architecture outlined in *Figure 1*, and a decision is made to re-route some of the IPTV traffic from SP1 to Residental Area A via the central node in the core network.

The delay experienced by data travelling from the service providers to the end-users, before and after the reconfiguration, is shown in *Section 4*.

4 Results

This section provides the results obtained from the experiment outlined in the previous section. We present the average delay, along with a moving average of delay for subscribers of SP1 in Residental Area A, before and after network reconfiguration has taken place following a change in network traffic conditions. 60 seconds after monitoring has begun SP2 decreases the number of services to subscribers in Residental Area A, the metric instrumentation reports this information back to the network management component. At this point the decision is made to route some of the traffic from SP1 to Residental Area A via the central node to ease congestion through load balancing between the upper and central components of the core network.

Figures 3 and 4 and below shows the instantaneous delay and average delay, respectively, experienced by all subscribers of SP1 in Residental Area A. Here we see subscribers to SP1 reporting a decrease in the delay they experience once the network modifications have taken place.

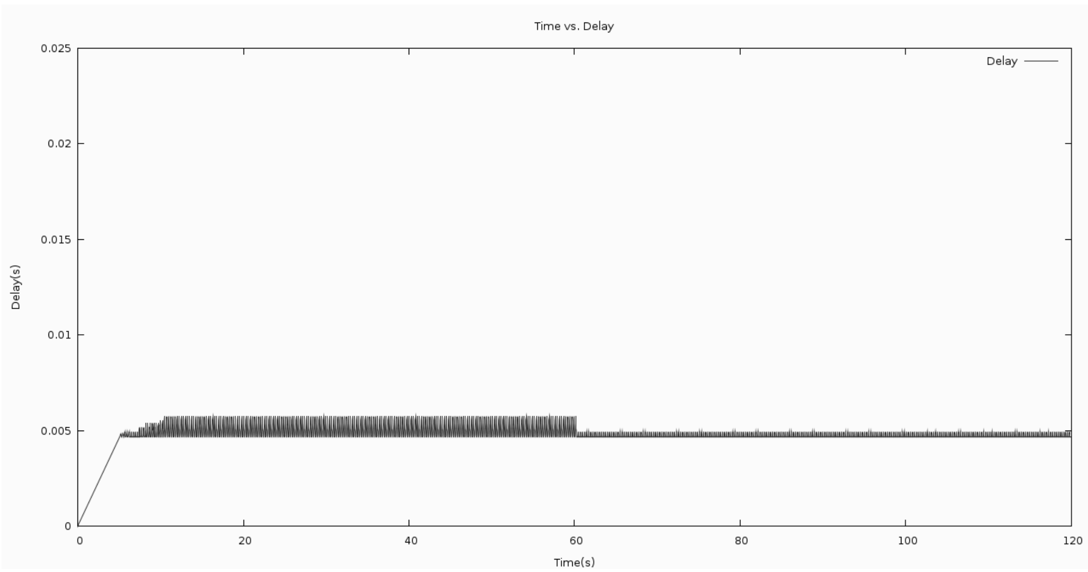

Figure 3: Instantaneous delay experienced by subscribers of SP1.

Figure 3 above shows the instantaneous delay reported by subscribers as the simulation progresses, initially as the IPTV traffic passes along the single outer link we see an intial reported delay across all subscribers (this is verified by *Figure 4*). As the network is reconfigured after SP2 decreases the number of IPTV streams to Residential Area A, and traffic from SP1 is load balanced across the outer link and through the central links, we see a decrease in the delay reported by the subscribers, as expected, along with a decrease in the variation among the reported delay values.

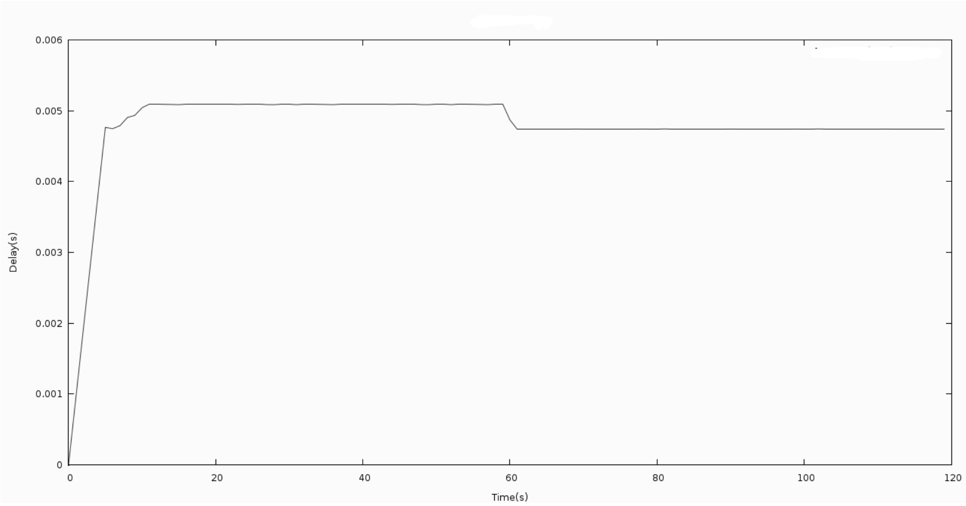

Figure 4: Average delay experienced by subscribers of SP1.

Figure 4 above shows the average of the reported delay values of the subscribers of SP1, mirroring the results shown in *Figure 3*. As the initial network configuration is used, we see only small variations of the average delay reported and upon reconfiguration of the network we see, as in *Figure 3*, the average reported delay decreasing by between 6.5% and 6.75%.

We also measured a moving average of delay experienced by, this takes into account the average delay experienced by the subscribers over a given window size. The window size chosen was approximately 1 second (depending on network conditions): this allows for generation of a metric that provides indications of past network conditions. This moving average of delay is shown below in *Figure 5*. Again we see a decrease in reported delay as before upon network reconfiguration.

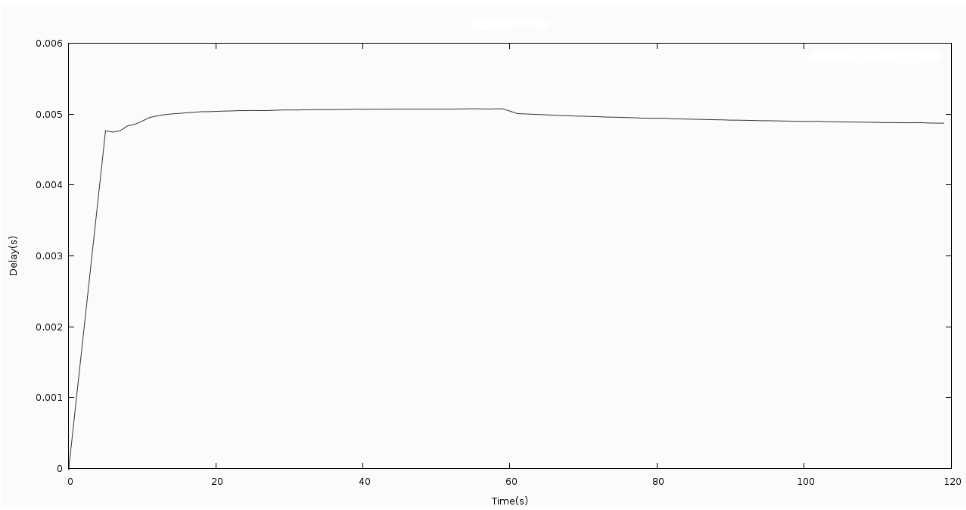

Figure 5: Moving windowed average of delay experienced by subscribers of SP1.

5 Conclusions

In this paper we described some of the QoS metrics that can be used to characterise network performance and how these metrics specifically apply to IPTV service provision. We also outlined a method to use these QoS metrics to provide information on current network conditions and how, using this information, a network management function can modify network parameters to optimise service flows through the network. Our simulation results show the decrease in delay experienced by subscribers of an IPTV service when using this approach, thus minimising excessive wait times and increasing end-user experience.

Future work will include extending the simulation setup to include multiple metrics and making decisions based on these, as well as storing previous network configurations for use as an aid to network reconfiguration. We also intend to investigate the impact of multiple service types concurrently using the network, as would be found in actual triple-play service deployments. Comparisons will also be carried out against other routing algorithms used in video content distribution.

Acknowledgements

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