Providing Core-Generated Feedback to Adaptive on Actively Managed Networks

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Abstract—**Next generation Internet comes with the responsibility to offer good support to maintain Quality of Service (QoS) to users that want to pay more to have better services. New management mechanisms are created to issue the sensibly augmented complexity of differentiated services. Multimedia applications are one of the main goals of this new infrastructure. This paper proposes a more flexible way to install and apply management mechanisms. This flexibility comes from a programmable management plane as structural component of protocol stack. With the capability to self-configure management actions, according to traffic situations, and with the possibility to build and install new mechanisms to help new services, an active and programmable network management can offer customized support to user applications. As a case study, we show a scheme where core nodes in a network deal with various management mechanisms, such as selective discard, feedback to source and priority queuing to achieve multimedia jitter-sensible transport.**

I. INTRODUCTION

The Internet is continuously experimenting changes. The network is becoming each time more powerful and popular all over the world. Information democracy is one of the reasons for its success. Technical influence behind this success is the increase of bandwidth availability inside network backbones (due to optical communication techniques, for example), associated with the development of new means for end users to access the network (e.g., ISDN, Cable Modems, ADSL), expanding the bandwidth upgrade to the edges of the network. Hence, human influence is due to a greater demand for services that can profit of this new and powerful infrastructure made available.

The success of Internet is leading to the creation of new services. Among these new possibilities are the Multimedia applications. There is an increasing availability in the Internet of services such as Live Radio and TV, Jukeboxes, On Demand Video Broadcasters, Video-Conferencing, Telephony, etc. These types of media are different from others because of their high requirements on Quality of Service (QoS) from the network to offer satisfactory results to users. For example, video-conferencing services are extremely sensitive to data transfer delay and its variation, while it is relatively tolerant to information loss. On the other hand, Jukebox services are not as exigent as videoconference in terms of transfer delay.

Although the Internet was not primarily designed to this kind of usage, its rapid popularization propitiates its broad use for these services. This incompatibility is due to the fact that Internet does not offer *a priori* definition and maintenance of Quality of Service to applications. Although this reality is in way of change after the deployment of new technologies like Service Integration (IntServ) with resource reservation protocols, such as RSVP (ReSerVation Protocol) [1] and Service Differentiation (DiffServ) [2], the best-effort philosophy over Internet have overcharged the end entities of the flow on the responsibility of QoS, flow and congestion control, because of the lack of processing power inside the network.

Furthermore, there is a strong trend to integrate various infrastructures such as fix and mobile telephony, wireless and Internet in one only infrastructure capable to offer customized support to different issues for services and its requirements. These services will follow specific tariff rules, once users are willing to pay more to have a better service. This new trend, based on IPv6 [3] and named "IPng (IP New Generation)" [4], is being studied and developed by research centers, industry and standardization organisms, such as IETF, to be the basis for the next generation Internet.

There is still work to do for QoS management to new generation Internet. A lot of lessons were passed from ATM (Asynchronous Transfer Mode) [5] and other first efforts to grant QoS. Some of these lessons were kept (label switching, reservation schemes, virtual paths, ABR, etc.) and other characteristics were left behind. The lack of internal mechanisms on the Internet took the responsibility for QoS control to the edges of the connections. Thus, source and destination must negotiate to control their communications.

On nowadays Internet, the end-to-end control for multimedia applications has being used as main tool for service designers to achieve acceptable QoS. However, this approach may overcharge the edges of the flow on processing and IO, and also cause waste of bandwidth and time, due to the usually excessive use of ACKs and signaling packets through all the pathway between source and destination. Nevertheless, another drawback of end-to-end control is the need to wait a RTT (Round Trip Time) to have source and destination synchronized to the actual flow situation. This problem may exist in ECN (Explicit Congestion Control) to IP [6], and EFCN (Explicit Forward Congestion Notification) to ATM [7].

This paper proposes the employment of activeness and programmability to enhance network management in new generation Internet. We argue that programmability can bring flexibility and proactivity to network management functions. Besides, activeness can help the deployment of autoconfiguration capabilities. As a case study, we evaluate the installation of some flow control and congestion management inside the network to control adaptive multimedia flows, and we observe the association of management mechanisms like feedback control, selective discarding and priority queuing to different situations of aggregate traffic.

The remainder of this paper is organized as follows. Section II gives a brief description of the proposed framework. On

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Section III, we show an environment where streaming multimedia traffic is submitted to different congestion and flow control mechanisms in different traffic loads. Section IV describes the experiments performed, and Section V discusses obtained results. Finally, Section VI presents our conclusions on the active management framework and on the experimentation performed, and the next steps of this research are discussed.

II. Active Management to QoS Internet

Even with the new capabilities that technologies like IPv6 and MPLS (Multiprotocol Label Switching) [8] have brought to the new generations of network infrastructures, there is still a great difficulty to develop and deploy new services. All new services must follow the formal procedures of standardization and commercial deployment to be available to users.

To solve these problems, DARPA-USA (Defense Advanced Research Projects Agency) has been investing on projects that aim to produce new active networking platforms [9], flexible and extensible at runtime to accommodate the rapid evolution and deployment of networking technologies [10].

Although this flexibility brought by active and programmable networks could help the development of new services, this technology could turn difficult the interoperabilization, because of the lack of standardization. Indeed, deployment of this completely open architecture over the network infrastructures of today has not yet been thought.

We argue that activeness and programmability have more smooth applicability on network management. These characteristics could make management task inside a carrier domain much more easy and flexible. Easy because management operations can be auto-configured and combined according to traffic situation, and flexible because the network is open to receive new management mechanisms. As a consequence, the network becomes more autonomous, offering capabilities that help self-healing and proactivity.

With the aim to achieve these capabilities, we propose the inclusion of a management plane to the Internet layer architecture. This kind of layer structure in three dimensions is not new. The B-ISDN (Broadband Integrated Digital Network) [11] reference model has a separate plane where all the management functions are implemented. Our proposal is different because we propose a management plane based on activeness and programmability. The new management plane will store legacy management mechanisms and will be able to receive new mechanisms by offering a Management API and an Execution Kernel. The proposed architecture for the Active Management Plane is shown in Fig. 1.

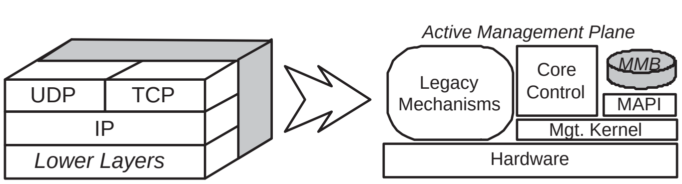


Fig. 1. Active Management Architecture.

This new plane is composed by the system hardware, that is shared between two entities: the set of Legacy Mechanisms and the Management Kernel. In the top of the Management Kernel, there are the Core Control, the Management Application Programming Interface (MAPI) and the Management Mechanism Base (MMB).

The Legacy Mechanisms are built-in mechanisms that are intrinsic to the network architecture chosen. Examples are DropTail, ECN, RED, etc. These mechanisms were chosen to be stored separately from customized mechanisms because as they are strictly inherent to the network architecture employed (*e.g.* IPv6, TCP, UDP, etc), they are already standardized and has no need to be taken away. As they are unchangeable, they can profit of a hardware optimized implementation and represent the core management mechanisms available.

The MMB is a persistent memory that stores the implementations of the customized management mechanisms made available to be used. MMB updates are completely controlled by the Core Control element and, consequently, by central carrier management entities.

The Management Kernel is an Operating System responsible for the creation and the processing of Execution Environments to the Core Control and to all management mechanisms that are built in the top of the MAPI and stored in the MMB. The Kernel controls local resources like queues, memory and processing power.

The role of the Core Control is to manage the execution of the running mechanisms, monitor network status and decide to, based on policies installed by the carrier network manager, swap running mechanisms with available idle mechanisms according to network status observed. The Core Control is also responsible to serve management information to carrier management system.

The API contains functions and procedures needed for management mechanism designers to have access to low level functions and resources.

III. Case Study: Adaptive Flow Control to Streaming Traffic

Network services experiment continuous evolution on network infrastructure potentials. As a consequence, new services are created and new requirements are stated. Multimedia services with dynamic media are examples of network applications that became popular in the Internet.

Dynamic media have a time dimension, and their meaning and correctness depend on the rate at which they are presented. Dynamic media include animation, audio, and video. These types of media have their intrinsic range of parameters where perception quality is adjusted. For example, to have a perceptually smooth movement, video must be played back at 25 frames (or 30 frames, depending on the video system used) per second. Similarly, when we play back a recorded voice message of music, only one playback rate is natural or sensible. Playback at a slower or faster rate distorts the meaning or the quality of the sound. Because these media must be played back continuously at a fixed rate, they are often called continuous media. They are also called isochronous media because of the fixed relationship between each media unit and time [12].

The way information is represented and transported through the network is an important characteristic of multimedia services. Streaming traffic is produced by multimedia applications, where destination initiates exhibition as soon as data begins to arrive. Synchronization between source and destination is necessary in order to keep destination buffer always containing data to be exhibited. Stream traffic contrasts with elastic traffic, which results from the transfer of documents (Web pages, files, MP3 tracks, etc.) using a transport protocol like FTP [13].

Multimedia services based on streams over the Internet use solutions that offer high level of data compression and end-to-end resource management and flow control. Flow control can be achieved by calibration of the media compression level – and consequently controlling media quality – with the aim to reduce or increase bandwidth consumption, according to its availability on the network.

These media types found a propitious operational infrastructure on ABR service class from ATM. On Internet, best-effort philosophy forces this kind of media to adapt itself to network conditions, and have an end-to-end flow control using as parameters indirect indications like packet loss.

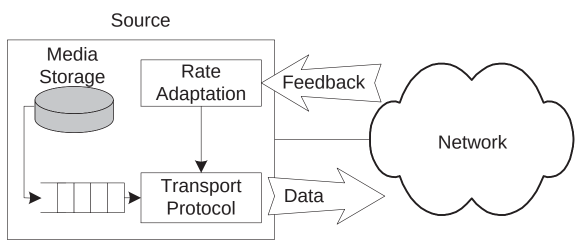


Fig. 2. Adaptive rate source.

Although applications can find advantages on taking responsibility of QoS control to inside the network, this is not possible in nowadays Internet, because routers do not have extra processing power (in addition to routing, basic resource management, for instance). Furthermore, there is not enough flexibility to implement new resource management mechanisms on the fly. Nevertheless, the inclusion of active-capable routers into Internet infrastructure, as proposed in [14], would offer the capabilities which are necessary to create these new services in even more intelligent fashion, permitting the discharge of control responsibility from end points of applications.

A. Related Work

The Real-time Transport Protocol (RTP) [15] is proposed to offer transport functions to real-time applications over unicast and multicast. This protocol is responsible only to transport multimedia data, not dealing with resource reservation or QoS maintenance. RTP is associated with Real-time Transport Control Protocol (RTCP) [15]. This protocol is responsible to monitor data delivery on large multicast networks and to offer a minimum set of end-to-end identification and control functionality. There are propositions of RTP payload format adapted to many media formats, such as MPEG-4 Streams [16], MPEG-1/2 Layer 3 (MP3) [17] and bundled MPEG [18].

One drawback of these approaches based on end-to-end control is that the responsibility of regulating transmission rate/quality is supported by applications on the flow end points. Although this can discharge processing overhead from network nodes, it may produce resource wasting and increase the delay to react to potential problems.

Zhang *et al.* [19] proposed a scheme where feedback congestion control system is extended from the endpoints to the routers in a MPLS domain.

Bush *et al.* [20] have created a scheme based on active networks to minimize end-to-end jitter by controlling the forward delay on intermediate nodes. Although they have shown good results in reducing end-to-end jitter, this solution increases sensibly the end-to-end delay.

Methods that exploits notification capabilities on intermediate nodes have been proposed. Floyd and Jacobson [21] proposed the Random Early Detection (RED). Its principle is based on active queue management, detecting incipient congestion by computing the average queue size. The gateway could notify connections of congestion either by dropping packets arriving at the gateway or by setting a bit in packet headers. When the average queue size exceeds a preset threshold, the gateway drops or marks each arriving packet with a certain probability, where the exact probability is a function of the average queue size. Other changes to this scheme were also proposed [22], [23].

Bouras and Gkamas [24] proposed a mechanism for monitoring the network condition and estimating the appropriate rate for transmission of multimedia data. In the same subject, Furini and Donald [25] have proposed a bandwidth allocation mechanism (BAM) that uses less bandwidth than peak rate BAM, while trying to provide the same service. This approach is applied only to stored video transmission.

Most management schemes are being designed to be deployed in networks cores that do not offer additional processing power other than routing. Generally, these networks have a fix set of mechanisms for congestion control and have little or no flexibility to combine mechanisms. Active and programmable networks are bringing a new paradigm to develop services that can profit of extra processing power inside the network. New management mechanisms with this profile must be developed.

IV. Experiments

This work shows the employment of different flow control and congestion management mechanisms adapted to various traffic situations, according to applications requirements. The main point of evaluation of this approach is that responsibility of monitoring is shared among intermediate nodes, which have autonomy also to trigger rate adaptation and preventive mechanisms to smooth congestion effects.

This experimentation is based on three distinct management mechanisms: Rate Adaptation with Feedback to Source, Selective Discard and Priority Queuing.

Rate Adaptation with Feedback to Source is used to readapt traffic characteristics to lack of bandwidth along the route. We introduce a feedback scheme that is partially controlled by internal nodes of the network, *i.e.* feedback signals can be produced by core nodes. We believe that feedback produced by the network and sent directly to the source can reduce the delay to rate adaptation.

Selective Discard is used as emergency mechanism to reduce congestion effects. This allows destination to continue to receive data, even if quality of this media is lower because of the discard. But the impact of packet loss in this case is minimized because of the discard of low priority packets. This scheme is based on a priority distinction among packets produced, assigned by the source. Priority decisions can also be defined inside the nodes, by discarding bigger packets, for example.

The third mechanism, Priority Queuing, is used to guarantee a faster and more reliable arrival of feedback signals to the source. This point is still subject to further experimentation.

The network is composed by 21 4-port active switches and 38 end systems that act as flow sources and destinations at the same time. All these entities are interconnected by links with capacity of 2 $Mbps$ and propagation delay of 10 ms. The active switches have output buffers of 200 packets.

Traffic sources generate packets with exponentially distributed size, with mean $\bar{S}_{pkt}$. We have defined 12 Service Levels (SL), inspired on MPEG 1 Layer 3 definitions. These levels are described in Table I. Hence, SL is determined by $\bar{S}_{pkt}$.

SL	Rate	SL	Rate
1	32 kbps	7	96 kbps
2	40 kbps	8	112 kbps
3	48 kbps	9	128 kbps
4	56 kbps	10	160 kbps
5	64 kbps	11	192 kbps
6	80 kbps	12	224 kbps

TABLE I
SERVICE LEVEL DEFINITIONS.

Packets can be classified as high priority data packet, low priority data packet, feedback packet or probe packet. Media data transport is separated into two different priorities in order to make selective discarding. Feedback packets have small size and carry feedback information from the network to the flow source. Probe packets are data packets that will trigger flow monitoring in the intermediate nodes. We choose to send this type of packet only periodically to avoid monitoring of every packet that will be excessively heavy on processing expense.

A source sends packets with mean time between departures $\Delta T_{def} = \frac{1}{F_{pkt}} s$, where F_{pkt} is the packet generation frequency (in packets per second). For our simulation, a data packet is generated as high priority with a probability of 7%. After each $NP - 1$ packets sent, the next will be marked as a probe packet. Thus, probe packet generation frequency is defined as $F_{pb} = \frac{1}{NP}$.

The exploited simulation environment is illustrated in Fig. 3.

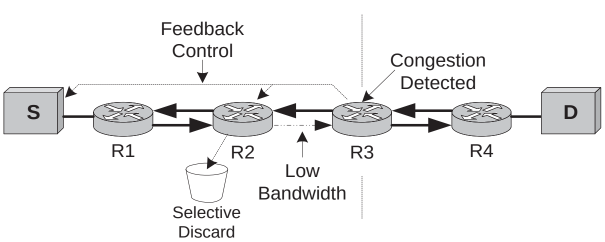


Fig. 3. Diagram of combined management mechanisms.

In the figure, the congested link on downstream way is located between nodes $R2$ and $R3$. When an abnormal situation is detected, the node where the event was detected will send a signal to his upstream node to trigger selective discard. This same message is forwarded to flow source, requesting transmission rate readaptation. The source will receive the feedback signal and must increase media compression in order to reduce transmission rate. The inner nodes will continue to monitor the flow and if abnormal situation persists, other feedback signals can be generated and sent to the source, requesting successive rate consumption adaptation.

In the performed experiments, destination is responsible for verifying and requesting the increase of transmission rate to the source by sending feedback packets. In this case, only the destination system can request increase of transmission rate. However, this task can also be distributed among network nodes, and we leave this adaptation to future work.

Inside the network, each node is responsible for monitoring network conditions by verifying its links and buffers, and flow behavior. Management decisions can be taken in response to events on the overall network status level, Class of Service (CoS) level or flow level. In our experiments, we have decided to adjust the granularity of control to a per-flow level. This requires node capacity to store flow state information. Detection can also be accomplished based on general attributes, like buffer occupancy, for example. This can help scalability and is previewed as future work.

Node status for each flow can be: *Normal*, *Exceeded* or *Waiting*. *Normal* status means that node is running with expected service level equals to actual service level. *Exceeded* status is signaled when a node receives a feedback generated by its downstream neighbor. This means that the delay inserted by the output queue of the current node was enough to exceed a defined delay tolerance. Thus, this node must be responsible for triggering selective discard as secondary management task in order to minimize effects while rate adaptation request is received by the source. *Exceeded* node also marks packets to notify downstream nodes. *Waiting* status is given to nodes that have received a communication of exceeded delay, and that are not the downstream neighbor of an *Exceeded* node. It means that the node has received the notification of its rate adaptation for a given flow, and it must wait for this adaptation. In this case, the node stops monitoring the flow until its rate acknowledgement is made. As rate adaptation is acknoledged and expected service level is equal to actual service level, node turns to *Normal* status.

Each flow is characterized by its original service level (SL_{orig}) and its expected service level (SL_{exp}). As rate adaptation to a given source is requested by a network node, its service level is decremented. Every node in the route must be notified. Upstream nodes are notified by the perception of a feedback signal in transit to the source, and downstream nodes are notified by receiving marked packets. All nodes stand waiting for flow adaptation, while *Exceeded* node operates with selective discard. When source adapts service level, all nodes in the path acknowledge service level change and node status is turned to *Normal*.

We detect abnormal situations by calculating the arrival delay of each probe packet. A maximum delay threshold was defined as a tolerance to arrival delay of a probe packet. If the difference between expected and actual arrival time is greater than this threshold, it is declared an "Exceeded Delay Event".

Two events are possible in relation to probe packet arrival. First, if the probe packet arrives with a delay (D_p) that exceeds a Decrementing Threshold Delay (ThD_{dec}), it is considered as an "excessive delay arrival" event, namely E_{dec}. If a probe packet arrives with delay lower than an Incrementing Threshold Delay (ThD_{inc}), it is considered as a "normal delay arrival" event (E_{inc}). The interval between ThD_{inc} and ThD_{dec} prevents normal variation on packet arrival time due to processing overhead (Fig. 4).

$$D_p \geq ThD_{dec} \longrightarrow E_{dec}$$

$$D_p \leq ThD_{inc} \longrightarrow E_{inc}$$

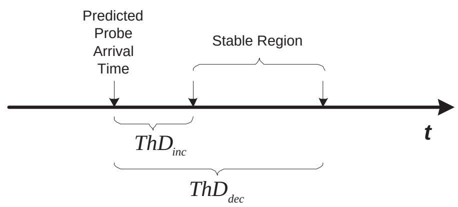


Fig. 4. Trigger parameters.

These two parameters are defined for each flow and are calculated in function of the difference between expected and actual service level and a given base delay tolerance ThD_b. Thus, each time stream rate is adjusted, the threshold to the next rate adaptation is also adjusted.

$$ThD_{dec} = (\Delta SL + 1) \times ThD_b \times \gamma$$

$$ThD_{inc} = max\{(\Delta SL + 1) \times ThD_b \times \gamma - \mu \times ThD_b; 0\}$$

Where $\Delta SL = SL_{orig} - SL_{exp}$ is the difference between expected and original service level to the flow. γ and μ represent the growing and stability coeficients. Growing coeficient defines the threshold variation between twoservice level changes. Stability coeficient stipulates the width of the stable region (Fig. 4).

To avoid mistaken decisions due to transient excessive delay situations, we use the reincidence of events rather than a single event to trigger management mechanisms. Status change is done by counting events. We define a counter Threshold (ThC). Inside the network, when a probe packet arrives and an event E_{dec} is declared, the event counter in increment. Otherwise, the counter is decremented. In the destination point, the decision to increase service level is also taken following a homologous criterium. If the event counter reaches the predefined counter threshold (ThC), management mechanisms are triggered.

In the destination, the decision to request service level incrementing is made following the same creteria used in internal nodes, taking into account the E_{inc} events.

A simple type of priority queuing is used to guarantee the arrival of probe and feedback packets. This policy defines the employment of a separate *Control Plane*, as used in ATM, to transport control and management signals.

V. Results and Discussion

Simulations were performed for 3 different network conditions: Low Charge, Medium Charge and High Charge. Average packet loss for these situations are 0.20%, 2.69% and 7.07%, respectively. These measures were performed for a plein best-effort network, without any traffic management mechanism.

Simulations take into account the impact of management schemes applied only to one flow. The results show the behavior of this flow submited to treatment. Impact of management schemes on every multimedia flows inside the simulated network was not yet evaluated. Results can be seen in Fig. 5.

Fig. 5(a), 5(b), and 5(c) show the average end-to-end delay observed on the concerned multimedia flow. In 5(a), we see that in situations of low charge, the association of feedback control and selective discard does not bring positive effects. On the other hand, this combination reduces signifantly end-to-end delay on medium and high charges situations, as shown in Fig. 5(b), and 5(c).

In Fig. 5(d), 5(e), and 5(f), we show the variation of jitter in destination. Jitter was calculated as proposed in the RTP protocol specification [15]. We can see that the employment of feedback has reduced jitter in all network situations tested. Feedback Control associated with Selective Discard produced good jitter reduction in all tested cases, even with packet losses that could influence jitter in destination. However, we observe that pure Feedback Control reached better jitter reduction than when associated with Selective Discard (Fig. 5(f)).

It can also be observed that feedback generated inside the network obtains better results than when feedback is produced by destination (End-to-End). This can be seen in Fig. 5(b)-(e). This effect can be due to the reduction of reaction time between congestion and rate adaptation when feedback signal traverse a shorter path to arrive in flow source.

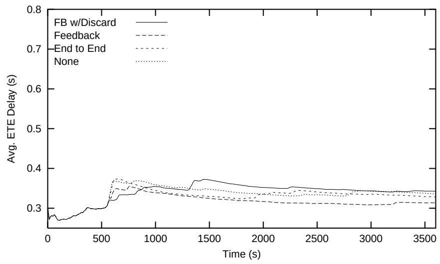
VI. Conclusions and Future Work

This paper proposes the employment of activeness and programmability to enhance network management task. The proposed arquitecture tries to profit of processing power inside the network to add flexibility and self-healing capabilities to management, while trying to keep transport functions independent of extra processing overhead.

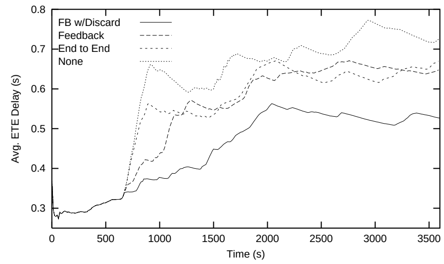
As a case study, we experiment the association of three management capabilities to solve flow and congestion control on multimedia stream-based applications. Feedback Control, Selective Discard and Priority Queuing were employed. Results show that the employment of a management schemes is dependent on type and severity of problems, and that mechanisms association can reach good results on solving management problems in different situations. We also see that Feedback Control based on internal nodes of the network can produce better results that when generated by flow destination (end-to-end).

Although this work is more focused on lower level management, we beleave that activeness and programmability can also bring good enhancements to higher level management (MIB management, Agent/Manager capabilities, Policy-based Management, etc.), and we leave this study to further work.

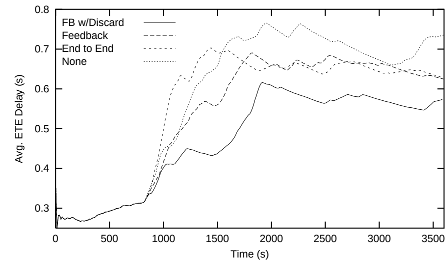
Several problems with the experimented schemes still exists.



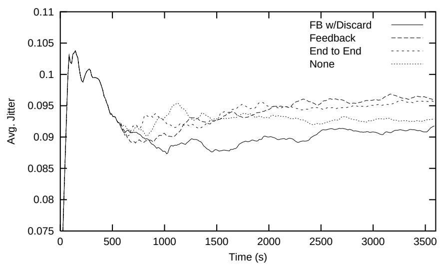
(a) Avg. ETE Delay (Low Charge)



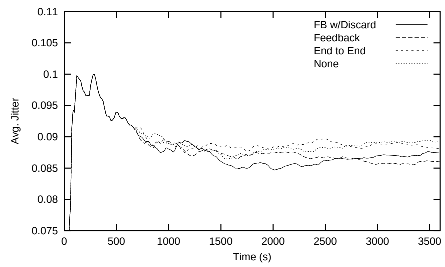
(b) Avg. ETE Delay (Medium Charge)



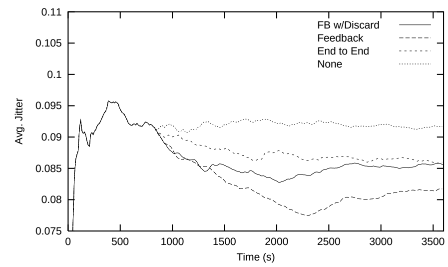
(c) Avg. ETE Delay (High Charge)



(d) Avg. Jitter (Low Charge)



(e) Avg. Jitter (Medium Charge)



(f) Avg. Jitter (High Charge)

Fig. 5. Simulation Results.

As an example, excessive state information produced by a per-flow granularity of control runs against scalability. To solve this problem, it is possible to use decision criteria based on the overall situation of the network.

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