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Combining quality of services path first routing and admission control to support VoIP traffic

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ABSTRACT

Existing interactive communication applications, like VoIP, are designed to manage losses and delay in real IP networks. We present a novel call admission control for VoIP based on blocking percentage calculation (BPC). Blocking rates were measured carrying VoIP for every router across the source–destination path. Ingress router selects the optimal path for the new request by calculating blocking rates across all source–destination paths. The proposed scheme dynamically adjusts the admission threshold to guarantee the voice quality of each flow. A simple algorithm design and evaluation results show that the proposed scheme is effective to achieve high utilization of network resources, while satisfying end-to-end targets in terms of blocking rate, delay, packet loss rate and fairness.

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1. Introduction

The Internet is considered as a foundation of a next generation network (NGN) [1,2]. The NGN is evolving in the direction of a packet-based network for both real time and non-real time traffic. Internet based NGN now is widely discussed in the standards bodies. The European Telecommunications Standards Institute (ETSI) [3] and the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) [4] have proposed the QoS architecture of NGN, in which the service stratum and transport stratum are provided. Voice over IP (VoIP) service, for instance, requires computational resources to meet its stringent delay when the connection is established by call setup signaling. Both the status of network resources and the QoS request from applications can be dynamically changed. Network resource control in the service stratum is performed during call setup signaling and the transport stratum reserves network resources for the QoS request.

The call admission control (CAC) mechanism is used to decide whether or not to admit a new traffic flow such that the previously admitted flows in a network still maintain their QoS requirements. CAC is essential for the QoS guarantee communications. The key component in ETSI NGN is the resource and admission control subsystem (RACS), which provides end-to-end QoS [3]. The control region of RACS covers the access network and the edge of the core network. In ITU-T NGN, access resource and admission control functions (A-RACF) make the admission decision based on the resource state of the access network, and the service-based policy decision function (SPDF) performs the policy-based decision and the control of the border of the core network [4].

An ideal CAC scheme must achieve the following objectives:

- Scalability: the scheme must scale well to large size networks to attain the required QoS guarantee.
- Robustness: the scheme must have the ability to handle against the failure nodes under various traffic characteristics.
- Efficiency: the scheme must be easily implemented and fast to achieve high link utilization.
- Fairness: the scheme must maintain the difference between the throughputs experienced by all the calls as small as possible.

We propose a new CAC algorithm to achieve these goals. We use the end-to-end measurement based admission control (EMBAC) in VoIP networks. Two types of measurement methods are considered in this proposal, namely blocking percentage calculation (BPC) and blocking percentage calculation with instant feedback (BPC-IF). We focus on controlling both average packet loss and end-to-end delay as a measure of QoS. The merits of this paper are threefold. First, our approach is extremely robust and is specially designed for VoIP traffic. The main shortcoming of probe-based approaches is that it may incur a setup delay, which is not acceptable for real-time traffic like VoIP. Unlike the probe-based approaches, our EAC-like (EAC will be described in Section 2) scheme uses the dynamic threshold value based on passive measurement of the blocking rate received from each node to make a decision to admit. Once a call request is admitted, the ingress selects one end-to-end path with lower blocking rate for incoming connection requests. The proposed scheme is robust. The calls in progress could be maintained even if one or more nodes

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fail. Second, the proposed mechanism is highly scalable since a small per-flow state is required to be maintained on the end-to-end path. The scheme uses passive measurement of blocking rate and only needs a few overheads to keep track of per-flow information. Third, the proposed scheme is highly utilized. Most previous researchers have employed a fixed threshold to decide whether to admit incoming connections into the network. The incoming flow is rejected when the network load exceeds a certain threshold. In the probe-based approaches, too many probes into the network at the same time can cause congestion. All of them may be rejected although some should be admitted. This leads to lower utilization. The proposed mechanism does not experience such problems because the admission threshold is dynamically adjusted based on feedback of failure rate of call requests. In both of the proposed algorithms (BPC and BPC-IF), a small spacing between the estimated admissible bandwidth and the real available capacity results in low blocking rate and, consequently, a higher utilization.

The rest of the paper is organized as follows. In Section 2, we describe the related work. Section 3 presents the proposed scheme. We show performance evaluation in Section 4. The simulation result demonstrates the feasibility of the proposed scheme. Finally, Section 5 concludes this paper.

2. Related work

In conventional CAC, one or several bandwidth brokers (BB) manage network resource based on the policy and take admission decisions for each call request [5,6]. However, this approach cannot scale well due to its centralized architecture. The ability of BB to handle growing volumes of flows suffers as the network size increases. Recently, many methods of scalable CAC have been proposed. We may categorize them into two main approaches: parameter-based [7–12] and measurement-based [13–28] CAC.

In parameter-based CAC, some traffic pattern and network topology usually are assumed in the network model and the aggregated resource consumption is required to be maintained below the total capacity. The accuracy of parameter-based CAC depends on the reliability of the assumed network model. Parameter-based CAC may result in low utilization, especially in the case of heavy load. The autonomous distributed control (ADC) approach [10] is regarded as one variant of parameter-based CAC. Rather than considering various flow characteristics and associated admission control procedures, ADC takes a specific traffic characterization and examines the computational aspects of performing admission control. The flow is characterized by piecewise linear envelopes and earlier deadline first (EDF) scheduling is made. However, the admission algorithm is highly computational and the whole architecture is not clear. The semi-Markov decision process (SMDP) can be used to develop optimum CAC under the condition of flow equality [11]. By using several feature parameters that represent the system or flow state, SMDP can obtain optimal evaluation regarding whether one arriving flow should be accommodated or not. A more efficient method to reduce the calculation cost of SMDP optimum CAC has been proposed [12]. The CAC aims to minimize the total call blocking rate as well as to maximize the number of flows accommodated. Moreover, it achieves almost the same performance as SMDP optimum CAC.

The measurement-based mechanism can be further classified into two approaches depending on the location of the admission decision. In the first approach, the admission decision is made by intermediate nodes as well as the coordination of intermediate nodes in the network [13–16]. In the second approach, endpoint admission control (EAC) [17–28] performs the admission decision based on the traffic condition of the network estimated by the end systems. EAC can be again divided into two categories: active [17–25] and passive CAC [26–28]. Active CAC performs the decision to join the network based on network probing. Probe-based admission control (PBAC) [17] provides a reliable upper bound to the packet loss probability a flow will suffer in the network. To perform the loss measurement, PBAC uses probe packets that are sent at the peak rate of the new session. PBAC also designs the queuing function of service class to differentiate data packets from probe packets, so that probes do not disturb ongoing data sessions. More design issues are derived for admissible conditions considering packet loss rate [18], link load [19], queue length [20] and delay [21–24] and even internet price [25] based on the available bandwidth, which is estimated through the egress by probing packets. Active CAC requires no explicit support from routers and thus avoids the scalability problem of maintaining the per-flow state at each router. However, the probing process might incur a rather long setup delay, especially in the case of bursty traffic. Thus, active CAC faces a scalability problem when the network utilization is high.

Passive CAC, on the other hand, uses passive measurement at the egress router to take the admission decision. In [26], a lightweight, scalable and distributed CAC for voice traffic has been proposed. Rate mismatches at the egress router is detected through passive monitoring of the average delay of n packets. Experimental results have shown that the proposed scheme is capable of delivering the stringent QoS requirements of voice traffic by regulating the end-to-end delay while achieving the satisfactory packet loss. However, the limitation is that it only works in the ISP control domain. Operating across multiple domains is still a challenging problem. A new QoS architecture, called a nonblocking network [27], has been proposed recently that requires no admission control inside the network and can still guarantee the congestion-free property. It applies to both IP-like and MPLS-like networks. In this architecture, as long as each edge node admits less than a specified amount of traffic, the network will never experience link congestion. A hybrid approach of combining explicit admission control and high-speed transport protocols to enable an opportunistic sharing of the capacity by flows in the Grid has been proposed [28]. The proposal provides a high acceptance probability of flows in the network while maintaining efficient network-resource utilization. Grid applications usually move large amounts of data between these resources within deterministic time frames. However, in most cases it is hard to specify the volume and the deadline in advance having heterogeneous bandwidth and delay requirements in real IP networks.

3. The proposed scheme

3.1. Searching algorithm

The proposed scheme uses the QoSPF [29] algorithm to compute QoS paths and select a path capable of meeting the QoS requirements of a given request. The architecture of QoSPF consists of three major components: the signaling component, the QoS routing component and the traffic manager. In this article, the resource reservation protocol (RSVP) is employed to signal from the source to the destination to reserve network resources. RSVP is resumed whenever the topology or routing path is changed. The traffic manager maintains the path assigned for use by a given request. The proposed scheme aims to provide a simple and scalable CAC, with minimal impact on the existing OSPF protocol and its current implementation. In QoSPF, network resource information is added and used to calculate QoS routes that can provide the resources needed for the flow. The resource information is advertised in Link Resource Advertisements (RES-LSAs) and Resource Reservation Advertisements (RRAs). In this article, we use the blocking rate as an indicator of availability of network resources. A RRA describes a router's reservations for a particular flow (source, destination) on its interfaces within an area. The RRA is used to indicate

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