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Traffic Engineering in the Voice Telephone Network

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Abstract

To engineer any telephone system, wireless or landline, we need some way of estimating telephony demand and criteria for satisfying that demand. Traffic engineering is the process of assigning facilities to satisfy a demand. Traffic engineering is particularly important inside the public switched telephone network (PSTN), the Internet, the cellular backhaul network, and other networks where demand from multiple users is carried over shared channels. Statistical methods can be applied to estimate demand and design facilities to meet peak demand. In this chapter we will explain those methods. First, we will look at the methods traditionally used in telephony, which assumed that all circuits were the same size and that the average length of telephone calls was unlikely to change very much. The results show that call capacity is most likely constrained by network devices rather than physical connections. Therefore, we recommend considering both packet throughput and bit throughput (bps) in determining the max call load. If network capacity is constrained by packet throughput, codec schemes would have almost no effect on the maximum call load.

Keywords: VoIP, Bandwidth, Traffic Engineering, Packet Throughput, Call Admission Control

1. Introduction

The key role of the IP traffic engineering is to maximize such that the network resources are used to the maximum while the all the performance requirements are satisfied.

Traffic engineering is the critical part in the IP intradomain operational network.

It can be used to address the operational network overall behaviour and the medium goals of the network involved.

It does not involve the issues of network dimensions like increasing the capacity of the network.

The framework used for calculating the weight of the link in a network where routing is done on the basis of shortest path has a crucial relationship in reference to Multi-Commodity Network Flow (MCNF). In this chapter we present the relation between the MCNF, Link Weight and the Shortest Path routing.

Further indicate how the IP traffic engineering issues can be considered as the issues of the MCSPRF (Multi-Commodity Shortest Path Routing Flow).

The commercial Linear Programming (LP) can be utilized to identify dual solution. This is very helpful for a network providers for whom the Meta heuristic Based Link Weight determination algorithm is not an option.

The growing popularity of Voice over IP (VoIP) is evident on the residential, enterprise, and carrier networks. The traditional IP-based networks are designed for data traffic, and there is no engineering consideration for voice traffic which is sensitive to packet delay and loss. To meet the new challenges of network convergence of both voice and data services on the same network, traffic engineering is important to network design as well as to the continual operation of the services. This paper provides an in-depth study of the VoIP traffic engineering and presents an enhanced traffic engineering model for VoIP.

2. Call Admission Control

The purpose of Call Admission Control (CAC) is to determine if the network has sufficient resource to route an incoming call. In the circuit-switched networks, the Call Admission Control algorithm is simply to check if there are circuits (or trunks) available between the origination switch and the termination switch. VoIP traffic is carried over packet switched networks, and the concept of circuits (trunks) is not applicable. However, the need for Call Admission Control (CAC) of VoIP calls is the same. Packet switched networks, by nature, accepts any packet, regardless of voice or data packets. When the incoming traffic exceeds the network capacity, congestion occurs. Control mechanism is needed to address the issue of congestion by traffic shaping, queuing, buffering, and packet dropping. As a result of this procedure, packets could be delayed or dropped. Delay is usually not an issue for data-only applications. Packet loss can also be recovered by retransmission, which is supported by many protocols, such as TCP or TFTP. However, retransmission would cause longer delay which is not acceptable to time sensitive applications. For voice traffic, delay and packet loss would degrade the voice quality, which is not acceptable to end-users.

The Erlang-B Model

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The Erlang-B model is the standard to model the network traffic of circuit-switched networks. It is known as the blocked-calls-cleared model [8], where a blocked call is removed from the system. In this case, the user will receive an announcement of circuit busy. Note that a busy announcement is not the same as busy signal, which is the case when the called is already on the phone. From the perspective of the Erlang-B model, not-answered-calls and busy calls are all considered successful calls. This section provides a brief overview of the Erlang-B model and its application to the circuit-switched network.

Traffic Measurement

In a circuit-switched network, the limiting resource is the number of circuits which is also known as trunks (N). The traffic load on the network is measured by Traffic Intensity which is defined as Traffic Intensity (A) = Call Rate × Call Holding Time where call rate is the number of incoming calls during a certain period of time. Call Rate is randomly distributed and follows the Poisson distribution. Call Holding Time is the summation of (a) call duration which is the conversation time, (b) waiting time for agents at call centre, and (c) ringing time. The measurement unit of Traffic Intensity is Erlang which is the traffic load of one circuit over an hour. For example if a circuit is observed for 45-minute of use in a 60-minute interval, the traffic intensity is 45÷60=0.75 Erlang.

The Model

The Erlang B model is commonly used to determine the mathematical relationship of the traffic measurements The assumptions of the Erlang B model are Infinite number of sources: The model implies that an infinite number of users who could make a call through the network. In practice, if the number of users is much larger than the number of trunks, this assumption is considered valid.

Random call arrival: Since we have a large number of users, each user may initiate or receive a call at any time. The call arrival is random and follows the Poisson distribution, which also implies that the inter arrival time follows the exponential distribution. The randomness also implies that call events are independent of each other, where Call[i] and call [i+1] are two independent calls.

Blocked calls are cleared: When a call is blocked due to insufficient resources (trunks), the user will get are cording or a fast busy tone. The call request is discarded (cleared) by the network and the user must hang up and try again at a later time.

Random holding time: The holding time (call duration and waiting time) also follows the exponential distribution. It should be noted that the assumptions of the Erlang-B model are transparent to the underlying networks, regardless of whether it is a circuit-switched network carrying traditional phone calls, or a packet switched network carrying voice calls in the form of VoIP. Another important note is that the Erlang B model has been proved to be fairly robust where minor violation of model assumptions would still yield useful and practical results for traffic engineering.

3. Voice over IP (VoIP) Networks

This paper studies three VoIP architectures:

Enterprise network

Access network of Internet service provider

VoIP carrier network

VoIP network for Enterprise

In the enterprise network, voice calls are carried over the packet-switched IP network within the enterprise. The VoIP network has an interface to the PSTN network, usually a T1 link. At the perimeter, the VoIP gateway provides the signalling interworking between Session Initiation Protocol (SIP) and Q.931/ISDN. The signalling function is to establish a duplex end-to-end connection between the caller and the called, and it could be initiated from either direction. After the call setup, the VoIP gateway extracts the voice payload from the IP packets (for outgoing calls) or encapsulates the voice payload onto the IP packets (for incoming calls). In some implementations, the enterprise phone network consists of IP phones, and a Call Manager. In other cases, the enterprise local phone system has both IP and analog phones. In the latter case, the call control process requires a hybrid PBX supporting both IP and analog calls.

Access Network

The access network, where an enterprise subscribes to the VoIP service through an Internet Service Provider (ISP).The VoIP traffic is carried over the public Internet which is a best-effort network and does not support QoS, we cannot apply Call Admission Control in this architecture. The engineering of trunks between the ISP voice gateway and the PSTN follows the Erlang-B model.

Tandem Service over a Carrier Network

The third VoIP architecture is tandem service over the carrier network.

The two major network elements are Voice Trunking Gateway and Soft switch. Voice Trunking Gateway receives Voice Time Division Multiplexing (TDM) traffic from legacy voice switches and converts it to IP packets and forwards the packets to the IP backbone for transport. Soft switch uses the Signalling System 7 (SS7) to interface with the legacy voice switches and also to interface with other soft switches. The purpose of the SS7 is to establish an end-to-end connection between the callers and called. It should be noted that the edge router may also accept VoIP traffic from another VoIP carrier.

4. VoIP Traffic Analysis

VoIP packets are transported over Real-time Transport Protocol (RTP) which in turn uses UDP. RTP provides sequencing and time-stamp to synchronize the media payload. Real-time Transport Control Protocol (RTCP) is used in conjunction with RTP for media control and traffic reporting. Our experiment shows that RTCP is only about 1% of the VoIP traffic, so RTCP traffic is excluded in our analysis for traffic engineering.

VoIP Packet Overhead

VoIP encapsulates digitized voice in IP packets. The standard Pulse Code Modulation (PCM) uses 256 quantization level and 8,000 samples per seconds. As a result, we have a digitized voice channel of 64 kbps (DS0).

If we use 20ms sampling interval, each sample will be 64,000 bps × 20 MS = 1,280 bits = 160 bytes this digitized voice is then encapsulated in an RTP/UDP/IP packet.

VoIP Traffic Characteristics

VoIP Systems use two types of messages on the IP networks:

Control Traffic

IP Voice

Payload Traffic. The control traffic is generated by the call setup and management protocols and is used to initiate, maintain, manage, and terminate connections between users. VoIP Control traffic consumes little bandwidth and does not require to be included in the traffic engineering modelling. It is possible to provision another overlay network for signalling messages which have more stringent requirements than the payload traffic. IP voice payload traffic consists of the messages that carry the encoded voice conversations in the form of IP packets. This type of traffic is what concerns network engineers as it requires relatively high bandwidth and has strict latency requirements. IP Voice payload Traffic is referred to as VoIP traffic and has some unique characteristics that require special handling and support by the underlying IP networks. The traffic characteristics that should be considered for VoIP networks are:

Real Time Traffic: Voice conversations are real time events. Therefore, transmitting voice data over IP networks should be performed as close to real time as possible, maintaining packet sequence and within a certain latency and latency variation (jitter) limits.

Small Packet Size: In order to minimize the sampling delay and hence maintain the latency constrains, VoIP data is carried in relatively small IP packets.

Symmetric Traffic: VoIP calls always generate symmetric traffic, same bandwidth from caller to call and from called to caller. This characteristic of VoIP traffic combined with the small packet size will have impact on the network devices

Any-to-any Traffic: any user might call any other user on the VoIP network which limits the ability of network engineers to predict the path of traffic flow. VoIP traffic might be initiated or terminated at any terminal point of the network, unlike many of the IP data networks where the majority of the traffic flows are known (e.g., clients to servers).

VoIP Call Requirements

Although human ear can tolerate some degradation in the voice quality and still be able to understand the conversation; however, there are certain requirements that should be met so that a VoIP call is acceptable. Transporting a Voice Call over the packet switched network has many challenges posed by the nature of the IP-based network which was originally designed for the data traffic. On the VoIP network, the major factors that determine voice quality are given as follows:

Delay: Represents the one-way end-to-end delay which is measured from speaker's mouth to listener's ear (mouth-to-ear). Delay includes coding/decoding, packetization, processing, queuing, and propagation delay. The ITU-T G.114 recommends for the one-way delay to be less than 150 Ms in order to maintain a quality conversation and transparent interactivity. If VoIP packets are delayed more than this limit, collisions might happen when the call participants talk at the same time.

Jitter: This is a measure of the variation in time of arrival (TOA) for consecutive packets. The original voice stream has fixed time intervals between frames; however, it is impossible to maintain this fixed interval on the IP network. The variation is caused by the queuing, serialization and contention effect of the IP networks. VoIP endpoints provide jitter buffers to compensate for the variation in TOA and to support the re-sequencing process. Packets enter the jitter buffer at a variable rate (as soon as they are received from the network) and are taken out at a constant rate for proper decoding. Buffering increases the overall latency and the jitter buffer size should be carefully chosen in a way to keep the overall latency (one-way delay) within the acceptable range. Packets arriving outside the jitter buffer boundaries will be discarded. Jitter calculations should also consider voice activity detection, out of order packets, and lost packets.

Packet Loss: Unlike data connections, VoIP has some tolerance to packet loss; however, if packet loss ratio exceeds a certain limit the quality of the call will be negatively affected. Several reasons might lead to packet loss in a network such as network congestion, transmission interference, attenuation, rejection of corrupted frames, and physical link errors. Different voice codec schemes have different tolerance to packet loss; however, it is recommends that packet loss be kept below 1%. It should also be noted that some packets might reach the intended destination and yet be dropped because they are late by more than the jitter buffer value. Therefore, measuring packet loss must also include the jitter buffer loss which is a factor of jitter buffer size and packet delay variation.

Vocoding (voice codec): the Vocoding scheme is another important factor in determining voice quality. A codec scheme could implement compression algorithm, redundancy and lost packet hiding techniques. Different Vocoding schemes also generate different digitally encoded voice frames in terms of frame size, bit rate, and the number of frames per second.

5. Conclusion

The Erlang-B model has been used by the telecom industry to determine the call capacity of circuit switched networks for many years. We are proposing to use the max call load for VoIP networks as a comparable measure to network trunks. With this modification, the Erlang-B model is applicable to determine the call capacity of VoIP networks.

Packet-switched networks, by nature, do not have the concept of blocking, and all incoming packets are accepted even if the new packets will add more loads on the network which could result in delay and packet loss. In the case of

VoIP, this will cause quality degradation to the new calls as well as to the existing ones. The solution to this problem is to use a Call Admission Control (CAC) where call manager or soft switch can apply the Erlang-B model to implement a CAC algorithm to accept or reject an incoming call request.

The traditional approach of calculating the maximum call load is based on network bandwidth, and our experiments show that this approach fails to work on some routed networks with high speed links. Our experiments show that packet throughput (pps) of network devices could be the constraint for VoIP traffic engineering. Based on our findings, network engineers should calculate not only the physical bandwidth of network interfaces but also the capacity of network devices.

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