On the Efficiency of Packet Telephony

Original

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1 Introduction

Abstract

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On the Efficiency of Packet Telephony
Networks

2 Guaranteed Services in Packet Switched Networks

Research are given in Section 5. Conclusions are drawn in Section 6.

Section 3 discusses the simulation models that were used to produce

Section 4 discusses the simulation models that were used to produce the results shown in Section 5. Conclusions are drawn in Section 6.

Section 2 discusses the simulation models that were used to produce the results shown in Section 5.

The paper is structured as follows: Section 2: Introduction; Section 3: Simulation Models; Section 4: The Methodology; Section 5: Results; Section 6: Conclusion.

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In a packet switched network, packets are independent data units which are transmitted exactly in the same order they arrive at the port.

2 Guaranteed Services in Packet Switched

2 Guaranteed Services in Packet Switched Networks

The network delay is stored in the switching element in the circuit.
This bound can be intuitively explained by considering that a burst of size $D$ is evenly distributed over a period of $\phi$.

$$D = \frac{\phi}{t_{\phi}} = \frac{\phi}{t} = \frac{D}{\phi}$$

By $\phi$-GPs, we mean that the aggregate burst over a period of $\phi$ is evenly distributed. The $\phi$-GPs algorithm is designed to provide a fair share of bandwidth among the sources, with each source receiving a share proportional to its demand. This is achieved by having each source send bursts of size $D$ over a period of $\phi$, where $D$ is the maximum permitted burst size.

The $\phi$-GPs algorithm operates on the following principle:

$$\frac{D}{\phi} = \psi$$

where $\psi$ is the minimum rate at which the network can operate. This ensures that the network is not saturated and maintain a fair share of bandwidth among the sources.

### 3.2 (Packet-by-packet) Generalized Processor Sharing

In the next section, we will discuss some details of the algorithms.
2.3 Evaluating the Efficiency of Guaranteed Services

Over Packet Networks

A major advantage of voice and video over packet networks is that they can theoretically be designed to offer guaranteed performance. This is in contrast to circuit-switched networks, which offer quality of service (QoS) guarantees through the use of prioritization and reservation mechanisms. In packet networks, QoS guarantees are realized through the use of admission control algorithms.

Admission control algorithms are used to ensure that the network can handle the traffic load without degrading the quality of service. These algorithms typically involve the following steps:

1. **Traffic measurement and probing**: Traffic characteristics are measured and probed to determine the traffic load.
2. **Admission control decision**: Based on the measured traffic characteristics, the admission control algorithm decides whether to accept or reject a new connection request.
3. **Resource allocation**: If a connection request is accepted, resources are allocated to the new connection.
4. **Quality of service monitoring**: The quality of service is monitored to ensure that the network can handle the traffic load without degrading the service quality.

The goal of admission control algorithms is to ensure that the network can handle the traffic load without degrading the quality of service. This is achieved through the use of resource allocation algorithms that dynamically adjust the network resources to meet the traffic demands.

In conclusion, admission control algorithms are essential for ensuring that packet networks meet the quality of service requirements of voice and video applications. These algorithms are designed to ensure that the network can handle the traffic load without degrading the quality of service, thereby ensuring a high level of reliability and predictability of the network performance.

2.4 Call Admission Control

The admission control algorithms must be able to handle the traffic load and ensure that the network can handle the traffic load without degrading the quality of service. This is achieved through the use of resource allocation algorithms that dynamically adjust the network resources to meet the traffic demands.

In conclusion, admission control algorithms are essential for ensuring that packet networks meet the quality of service requirements of voice and video applications. These algorithms are designed to ensure that the network can handle the traffic load without degrading the quality of service, thereby ensuring a high level of reliability and predictability of the network performance.
3.1 Call Duration Model

The simulation scenario is also introduced in section 3.1 of the simulation model.

The purpose of this section is to describe the model of the simulation network.

The model is composed of three main components: the call initiation, call processing, and call termination. The model is designed to simulate the behavior of the network with different network conditions and traffic loads. The model is also capable of handling multiple calls simultaneously, allowing for a more realistic simulation of network behavior.

The simulation environment is used to study the behavior of the network under different conditions. This includes parameters such as call arrival rate, call duration, and network load. The simulation environment allows for the visualization of network performance metrics such as call blocking probability, call completion probability, and network utilization. The simulation environment is also used to evaluate the performance of different network configurations and configurations.
Figure 1: Probability density of call duration as generated by the simulator.

3.2 Voice Encoding

Functions

In a production model, we use the weighted composition of a more accurate model in which the call duration is distributed according to a normal distribution. Because of new and different traffic patterns, the model proposed in [6] focuses on places with a high density of call start and end locations. The exponential model was chosen in the early days of voice transmission (TE), and phone calls are modeled as Poisson processes.
3.4 Call Admission Control

The major difference between admission control for fixed punching and that for mobile punching is the overhead involved in the first step. This overhead is incurred in the admission control function and can be divided into two parts:

1. **Network Overhead:** This includes the time required for the network to process the call request.
2. **Transport Overhead:** This includes the time required for the transport network to process the call request.

In general, the network overhead is much smaller than the transport overhead. Therefore, the overall time required for the admission control function is dominated by the transport overhead.

**Algorithm:** The algorithm for call admission control can be described as follows:

1. **Check if the number of calls exceeds the maximum limit.**
2. **If the limit is exceeded, reject the call.**
3. **Otherwise, accept the call.**

The admission control function also includes a mechanism to handle overload conditions. This mechanism can be activated if the number of active calls exceeds a certain threshold.
3.6 Network model

The mean time is calculated over the simulation (or the number of samples) and the median is calculated over the distribution of the values. The mean is not affected by extreme values, while the median is.

The network model is used to simulate the performance of the network. The network model is a mathematical model that describes the behavior of the network.

The performance of the network is determined by the network model. The network model is a mathematical model that describes the behavior of the network. The network model is used to simulate the performance of the network.
4 Simulation Results

Figure 3: Example from the Topology of a Circuit Switched Telephone Network

Figure 4: Network Topology used in the simulation.
4.2 Packetization

Packetization is the process of converting data into packets, which are then transmitted over a network. Packetization plays a crucial role in the efficient and effective transmission of data over a network. In this section, we will explore the principles and implications of packetization in detail.

As mentioned earlier, the packet format is an integral part of the protocol stack. The packet format defines the structure and content of a packet, which are essential for the correct interpretation and processing of the packet by the recipient.

The packet format includes several fields such as the header, data payload, and trailer. The header contains information about the sender, receiver, and the type of data being transmitted. The data payload contains the actual data being transmitted, while the trailer includes information about the end of the packet.

Packetization is crucial for efficient network operation. It allows for the effective use of network resources and ensures the timely delivery of data. By dividing data into packets, network protocols can efficiently manage network traffic and ensure that data is delivered in a timely and error-free manner.

In conclusion, packetization is a fundamental concept in network communication. It plays a critical role in ensuring the efficient and effective transmission of data over a network. Understanding packetization is essential for anyone working in the field of networking and communication.

4.3 Bandwidth Over-allocation

Bandwidth over-allocation occurs when the allocated bandwidth is exceeded by the actual data transmitted on the network. This can happen due to various reasons, including network congestion or over-estimation of the required bandwidth.

The impact of bandwidth over-allocation can be significant, leading to increased latency, decreased performance, and potential loss of data. It is crucial to monitor and manage bandwidth effectively to avoid over-allocation and optimize network performance.

Effective bandwidth management involves balancing the need for sufficient bandwidth with the limitations of the network infrastructure. This requires a careful analysis of network usage patterns and the implementation of appropriate measures to ensure efficient bandwidth utilization.

In summary, bandwidth over-allocation is a key consideration in network management. By understanding the principles of bandwidth management and implementing effective strategies, network administrators can ensure optimal performance and reliability.

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**Diagram:**

[Diagram illustrating Over-allocation Overhead (Packet Delay 32ms)]

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**Figure 2:** Efficiency indices on the 70% - 10% with high packetization.

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**Table:**

<table>
<thead>
<tr>
<th>Over Load</th>
<th>Over allocation Overhead (Packet Delay 32ms)</th>
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<tbody>
<tr>
<td>Efficient</td>
<td>Improvements in the 70% - 10% with high packetization.</td>
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</table>
Offered Load (Erlang)

| 9.3 | 27.8 | 46.4 | 64.9 | 83.4 | 102.0 | 120.5 | 139.1 | 157.6 | 176.2 | 194.7 |

20% 40% 60% 80% 100%

Link Occupancy (%)

Circuit Switching (PCM64)

Pack Delay 32 ms

Pack Delay 18 ms

Voice over IP: Link Performance

Voice over IP: Overallocation Overhead (Pack Delay 18 ms)

4.2.1 The Payload

Utilization Efficiency

The efficiency of the link over IP is lower than that over IP due to the overheads associated with the protocol headers. This section presents a computational model for the efficiency of the link over IP.
Figure 6: Blocking Probability and Real Load.

Figure 7: Blocking Probability and Effective Load.

Network Efficiency: the Real Load

Network Efficiency: the Effective Load

The network efficiency is defined as the overall network efficiency multiplied by the call blocking probability. This is the ratio of the number of calls accepted to the total number of calls attempted. The call blocking probability is defined as the ratio of the number of calls blocked to the total number of calls attempted.

The difference between the two is the effective load. This is the load that is actually being carried by the network. The real load is the actual load being carried by the network, while the effective load is the load that is being blocked.

In the case of real-time services, the effective load is usually much lower than the real load, because the blocking probability is very high. This is because the network is not able to handle the high traffic load, and so it has to block a large number of calls.

In the case of stored service, the effective load is usually much lower than the real load, because the blocking probability is very low. This is because the network is able to handle the high traffic load, and so it is able to accept a large number of calls.

The network efficiency is an important metric for characterizing a network. It is used to compare the performance of different networks, and to determine the appropriate level of service to provide.
Packet Delay (ms)

29.5
25.0
23.5
22.0
17.5
16.0
13.0
10.0

Circuit Switching (ADPCM32)

20000 40000

Circuit Switching (PCM64)

60000 80000

Voice over IP over

120000

Voice over ATM

140000

Packetization Overhead: the Real Bandwidth

The header size depends on the protocol and on the network, which is derived from the previous section.
4.3 SONET/SDH and Voice Compression

The primary objective of the SONET/SDH system is to provide a reliable, high-speed, high-quality voice service over a single fiber optic cable. This is achieved by using a time-division multiplexing technique, which allows multiple voice channels to be transmitted simultaneously on a single fiber optic cable. Each voice channel is assigned a specific time slot, and the information from each channel is transmitted during its designated slot.

The SONET/SDH system uses the ITU-T G.821 and G.822 standards for digital signal transmission. These standards define the physical layer and the multiplexing structure of the SONET/SDH system. The SONET/SDH system is designed to be compatible with other digital transmission systems, such as the Digital-Synchronous Carrier (T-carrier) system, which is used in the United States.

The SONET/SDH system uses a 2.048 Mbps channel to carry voice signals. Each voice channel is encoded using a digital modulation technique, such as pulse code modulation (PCM). The encoded signals are then multiplexed together to form a single optical signal, which is transmitted over a single fiber optic cable.

4.4 The Optimal Packet Size

Packet Delay (ms)

The optimal packet size is determined by the trade-off between the delay and the number of packets that can be transmitted over a single fiber optic cable. The delay is determined by the time required for the packets to travel from the source to the destination. The number of packets is determined by the available bandwidth of the fiber optic cable.

The optimal packet size is determined by the following formula:

\[
\frac{1}{1 + \frac{DT}{PRT}} = \frac{1}{1 + \frac{2 \times 10^{-9}}{10^{-6}}} = 0.004
\]

where

- \(DT\) is the delay introduced by the network
- \(PRT\) is the packet round-trip time
- \(P\) is the packet size
- \(R\) is the available bandwidth of the fiber optic cable

The optimal packet size is determined by minimizing the delay introduced by the network, while ensuring that the number of packets that can be transmitted over a single fiber optic cable is maximized.

The optimal packet size for voice transmission over a single fiber optic cable is typically in the range of 64 to 256 bytes. This ensures that the delay introduced by the network is minimized, while still providing sufficient bandwidth for the transmission of voice signals.
Discussion

5. Impact of Prediction Delay on the Link Efficiency

![Graph showing the impact of prediction delay on link efficiency](image)

The equation for $\frac{I}{H}$ is given by:

$$\frac{I}{H} \sim \frac{D_{track}}{D_{reg}}$$

where $D_{track}$ is the delay experienced by the track and $D_{reg}$ is the delay in the regeneration process. The optimal packet size depends on many parameters. However, a common formula for the optimal packet size is:

$$\frac{I}{H} = \frac{D_{track}}{D_{reg} + \text{Prop}}$$
References

Acknowledgments

The main conclusion we can draw from the simulation results is that...
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