On the Efficiency of Packet Telephony

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Introduction

Abstract

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On the Efficiency of Packet Telephony
networks

2 Guaranteed Services in Packet Switched Networks

The paper is structured as follows: Section 2 discusses the motivations, Section 3 describes the mechanisms, Section 4 examines the implementation, and Section 5 compares the performance of a packet switched network and the Internet. The paper is organized as follows: Section 2 discusses the motivations, Section 3 describes the mechanisms, Section 4 examines the implementation, and Section 5 compares the performance of a packet switched network and the Internet. The paper is structured as follows: Section 2 discusses the motivations, Section 3 describes the mechanisms, Section 4 examines the implementation, and Section 5 compares the performance of a packet switched network and the Internet.
This section describes in some detail the two optimizations which allow the GPs to determine the best packet order for transmission according to their bandwidth constraints. The main difference between the two optimizations is that the first one assumes that all packets are transmitted in order, while the second one allows for a more flexible packet scheduling. The second optimization is more suitable for real-time applications, such as video streaming, where delays must be kept to a minimum. The first optimization is better suited for applications where delays are less critical, such as file transfers.
2.3 Evaluating the Performance of Guaranteed Services

Over Packet Networks

The primary benefits of packet networks are that they can accommodate the traffic requirements better than circuit networks, and they can be used to support a wide range of services. However, packet networks also have some disadvantages, such as the possibility of congestion and the need for more complex network management. In addition, the performance of packet networks can be affected by the traffic patterns and the characteristics of the underlying network.

In the context of guaranteed services, packet networks can be used to provide a certain level of performance guarantee. This can be achieved through the use of traffic management techniques, such as traffic policing, traffic shaping, and traffic filtering. These techniques can be used to control the amount of traffic that is transmitted over a network and to ensure that it meets certain performance requirements.

2.4 Call Admission Control

Call admission control is a mechanism for controlling the amount of traffic that is transmitted over a network. It is used to prevent congestion and to ensure that the network can handle the incoming traffic. Call admission control is typically implemented at the network layer and is based on the principles of queuing theory.

The admission control process involves the following steps:

1. The network receives a request for a call from a user or a service provider.
2. The network checks whether the call meets the admission criteria.
3. If the call meets the admission criteria, the call is admitted to the network.
4. If the call does not meet the admission criteria, the call is rejected.

The admission control process is typically implemented using a combination of traffic management techniques, such as traffic policing, traffic shaping, and traffic filtering. These techniques are used to control the amount of traffic that is transmitted over a network and to ensure that it meets certain performance requirements.

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3.2 Call duration model

The simulation scenario is also introduced.

The figure shows a diagram of a call model, with each node and connection illustrated. The model includes various parameters and is used to simulate the behavior of calls within the network. The model incorporates the effects of different factors, such as call duration, to accurately reflect the real-world scenario.
Figure 1: Probability density of cell duration as generated by the simulator.

3.2 Voice Encoding

Simulators are no proposed by the component model, and if (x) is produced by the simulation, the probability density is that of the component model. Therefore, the probability density function is the product of the probability density functions for each of the components, i.e.,

\[ f(x) = f(x_1) \cdot f(x_2) \cdot \ldots \cdot f(x_n) \]

Functions in a probability distribution obtained by the weighted composition of a more accurate model in which the cell duration is distributed according to these functions. This model is not a perfect representation of the data and may not always be effective, especially for short calls and in the emergency process. This erroneous simple model was derived in the early days of simulation and is not recommended as a P-voice simulation.
3.4 Call Admission Control

The Call Admission Control (CAC) algorithm is used to control the admission of new calls into the system. The CAC algorithm helps to ensure that the network resources are not exceeded, and that the quality of service for existing calls is maintained.

The CAC algorithm is based on the following principles:

1. The maximum number of calls that can be admitted is determined by the available resources in the network.
2. Each call is assigned a specific priority level based on the type of service it requires.
3. The CAC algorithm uses a mathematical model to calculate the impact of each new call on the network resources.
4. If the impact of a new call would exceed the available resources, the call is rejected.

The CAC algorithm is important because it helps to prevent congestion and ensure that the network remains stable. By limiting the number of calls that can be admitted, the CAC algorithm helps to maintain the quality of service for all users.
network model

Network model

3.6 Network model

4.3 Statistic module

In order to equalize the measurement results obtained in the simulator, a statistic module is included in the simulator. The efficient indices described in Section 2.3 are measured by simulating.

The statistic module is located at the network interface. This is the only connection implemented in the simulator.
4 Simulation Results

Figures 1 and 2: Network topologies used in the simulation.

Figures 3 and 4: Excerpt from the topology of a Circuit Switched Telephone Network.
4.2 Packetization

When compared to the next section, the difference between the packetization and protocol overhead are highlighted. However, this is not the case for the overhand of the protocol. The difference between these is less significant in understanding the real difference between the packetization overhead and protocol overhead.

In the figure, "Packetization of the packet" shows the difference between the received packet and the original packet. The difference is expressed as a percentage of the original packet size. The "Packet Overhead" shows the efficiency of the packet overhead compared to the original packet size. The "Efficiency Index" indicates the percentage of the original packet size that is affected by the packet overhead.

Overall, the packetization overhead is a good representation of the processing load on the network. The delay introduced by the protocol overhead is lower than the packetization overhead, which is consistent with the overall network performance.

4.3 Bandwidth Over-allocation

The figure "Efficiency Index of the TCP - 1.0" with high packetization shows the efficiency index of the transport protocol under different load conditions. The protocol efficiency is represented by the curve. The "Over-allocation Threshold" is the point where the efficiency index reaches 100%. The "Effective Load" shows the actual load on the network, and the red load curve is the performance of the transport protocol under the given conditions.

Under the assumption that the network performance is good, the efficiency index of the transport protocol is shown to be high. However, when the network performance is poor, the efficiency index drops significantly. The performance of the transport protocol is also affected by the packetization overhead and the protocol overhead. The efficiency index is calculated as the ratio of the actual transmission time to the ideal transmission time.
Offered Load (Erlang)

Effective Load

Payload: Effective load over the effective load.

Payload: Impact of packet size over the effective load.
The network efficiency is considered low. The blocking probability and real load, shown in Figure 6, indicate that the network is subject to congestion. The high blocking probability is due to the limited number of available paths in the network. The real load, on the other hand, is relatively low compared to the maximum capacity.

Figure 6: Blocking probability and real load.

Network Efficiency: The Real Load

Figure 7: Blocking probability and effective load.

Network Efficiency: The Effective Load

The network efficiency can also be measured by studying the call blocking probability. Figures 8 and 9 show the blocking probability for different call arrival rates. The blocking probability increases as the call arrival rate increases, indicating that the network is becoming congested.

Figure 8 and 9: Call blocking probability for different call arrival rates.

The efficiency of the network can be improved by implementing additional routing strategies or by increasing the number of available paths.
Packet Delay (ms)

29.5
25.0
22.0
20.5
19.0
16.0
13.0
8.5
7.0
0

Real Bandwidth (byte)

Circuit Switching (ADPCM32)
Circuit Switching (PCM64)
Voice over ATM
ATM
Packetization Overhead: the Real Bandwidth over the Pseudo Bandwidth of a Phone

The header size depends on the protocol and packetization overhead in the network.
4.4 The Optimal IP Packet Size

The optimal IP packet size is determined by the need to minimize the delay and the number of packets transmitted over a network. The equation for the optimal packet size is given by:

\[
I^o = \frac{R \cdot (D + 2\cdot D_0)}{B_0}
\]

where:
- \( I^o \) is the optimal packet size
- \( R \) is the data rate in bps
- \( D \) is the delay in seconds
- \( D_0 \) is the delay due to protocol overhead
- \( B_0 \) is the bandwidth in bps

The optimal packet size is the key parameter in determining the efficiency of the network. A smaller packet size reduces the delay but increases the overhead, while a larger packet size decreases the overhead but increases the delay. The optimal packet size is determined by finding the balance between these two factors.

4.3 SONTF/SDH and Voice Compression

Voice over IP (VoIP) is becoming increasingly popular as a cost-effective alternative to traditional circuit-switched voice services. VoIP allows voice traffic to be transmitted over the internet, reducing the cost of long-distance calls. However, VoIP is not without its challenges. One of the main issues is the potential for packet loss, which can result in echo or other degradation of the voice quality.

To address this issue, voice compression algorithms are used. These algorithms reduce the amount of data required to transmit voice traffic, thereby reducing the number of packets required to transmit a call. This not only reduces the potential for packet loss but also reduces the bandwidth required to transmit the call.

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The equation for the impact of packet loss on the perceived quality of speech is given by:

\[
Q = 0.1 \cdot I^o \cdot P
\]

where:
- \( Q \) is the perceived quality of speech
- \( I^o \) is the optimal packet size
- \( P \) is the packet loss rate

This equation shows that the perceived quality of speech is directly proportional to the packet loss rate, and inversely proportional to the optimal packet size. Therefore, reducing the packet loss rate and increasing the optimal packet size can improve the perceived quality of speech.

The optimal packet size is determined by finding the balance between the delay and the number of packets required to transmit the call. This is done by performing simulations and experiments to determine the optimal packet size for different network conditions.
Packetization Delay (ms)

\[ T = \frac{1}{2} + \frac{H}{2} \]

Discussion

Figure 12: Impact of Packetization Delay on the Link Efficiency

The optimal packet size depends on many parameters. However, a common equation for peak load is given by:

\[ \text{Peak Load} = \frac{1 + H}{H} \text{Ideal Load} \]

This equation is a good approximation when the link has high capacity.

\[ \text{Optimal Pack Delay} \approx \text{Ideal Pack Delay} \cdot \frac{1 + H}{H} \]

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When accounting for some of the delay components in (9), the optimal packet size depends on many parameters. However, a common equation for peak load is given by:

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This equation is a good approximation when the link has high capacity.

Packetization Delay (ms)

\[ T = \frac{1}{2} + \frac{H}{2} \]
References

Acknowledgments

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Transactions