On the Efficiency of Packet Telephony

Original

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Introduction

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On the Efficiency of Packet Telephony
networks.

2 Guaranteed Services in Packet Switched Networks

In a packet-switched network, the transport service is provided at the 
packet level. The network is divided into a set of interconnected nodes, each 
acting as a router or a switch. Each node is responsible for forwarding 
packets to their destination. The network operates by sending packets 
through the network, with each node determining the best path for the 
packet to reach its destination. The network uses routing protocols to 
determine the path for each packet, ensuring that it reaches its intended 
destination. This allows for a high degree of flexibility in network 
design, as packets can be sent over a variety of routes, providing 
reliability and fault tolerance. The network is designed to handle 
bursty traffic efficiently, allowing for a high level of scalability and 
fault tolerance. This makes packet-switched networks ideal for 
applications that require a high level of reliability and 
flexibility.
This bound can be interpreted as the access node delay, 

\[ D = \frac{\beta_0}{\beta_0 - \lambda} \]

where \( \beta_0 \) is the minimum service interval, and \( \lambda \) is the offered load. The access node delay is defined as the time it takes for a packet to be transmitted from the access node to the corresponding service node.

The approach is based on the assumption that the access node delay is the dominant factor in the overall system performance. The service node delay is considered negligible compared to the access node delay.

\[ D = \frac{\beta_0}{\beta_0 - \lambda} \]

By calculating the average access node delay, one can determine the performance of the system under different traffic conditions.

To improve the system performance, various techniques can be employed, such as load balancing, prioritization, and queueing disciplines.

This approach is particularly useful in network systems where the access node delay is a critical factor in determining the overall system performance.
1. Real-time efficiency takes into account the amount of real-time traffic that flows through the network, which helps to ensure that important data is transmitted promptly.

2. Over Packet Networks

The difference in how the CVC is handled over packet networks is significant. While the CVC is handled in a more straightforward manner in circuit-switched networks, it requires additional processing in packet networks to ensure reliable delivery of data.

3. Call Admission Control

The CVC admission control mechanism is crucial for maintaining quality of service in packet networks. It ensures that only calls that meet the network's capacity constraints are admitted.
3.1 Call Duration Model

The simulation scenario is also introduced. The rest of the simulation scenario is described in more detail in the simulation module. The call duration model describes the outcome of each call and generates both the call handling behavior. The call handling behavior of each call can be accepted or rejected. If accepted, a new call is generated and the process continues. If rejected, the call is ended.

We define a set of four efficiency parameters that are independent from the network. These parameters are used to evaluate the efficiency of the network.

The simulation environment

The higher the fraction of new transmission capability supported by the network, the higher is the fraction of new transmission capability supported by the network. The same holds true for the fraction of existing transmission capability.

The effective load factor is the ratio between the number of calls offered to the network and the total number of calls offered to the network.

The call blocking probability is the ratio between the number of calls rejected and the number of calls offered to the network.

The effective load is the product of the effective load factor and the effective load factor (more)

To evaluate the efficiency of the network, we use the following parameters:

1. The effective load factor is the ratio between the number of calls offered to the network and the number of calls offered to the network. The same holds true for the fraction of existing transmission capability supported by the network.

2. The call blocking probability is the ratio between the number of calls rejected and the number of calls offered to the network.

3. The call duration model describes the outcome of each call and generates both the call handling behavior. The call handling behavior of each call can be accepted or rejected. If accepted, a new call is generated and the process continues. If rejected, the call is ended.

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Simulink C code readable.pdf

1. Probability density of call duration as generated by the simulator.

2. PCDM (Packet Continuation Delivery Methodology)

3. Voice Encoding

functions

simulink code readable.pdf

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functions
3.4 Call Admission Control

Conventionally, the most common physical layer transponders in the recombiner are the Precoordinated Digital Hierarchy (PDH) and the 52-Mbps Digital Hierarchy (DSH). We consider these two cases.

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3.3 Link model and Protocol Stack

A protocol stack is a set of rules and formats that define how data is communicated between devices. Each layer of the stack is responsible for a specific aspect of data transmission.

The stack is typically divided into layers, each with its own set of protocols and functions. The lower layers are responsible for basic transmission, while the higher layers handle more complex functions such as error correction and network management.

For example, the Transport Layer (TCP) is responsible for ensuring that data is delivered reliably, while the Application Layer (HTTP) is responsible for the end-user interaction.

In summary, the Link model and Protocol Stack provide a framework for ensuring that data is transmitted efficiently and reliably between devices.
network model

Conversely, the statistical model determines when the simulation can be stopped since the performance measures are observed during the simulation.

The network model is selected over the statistical model because the network model is more accurate and provides more detailed results. The statistical model is used to determine the performance of the network. The network model is more complex and requires more computational power than the statistical model. However, the network model provides more accurate results. Therefore, the network model is used to determine the performance of the network.

The network model is used in this simulation.
4 Simulation Results
4.2 Packet Over-allocation

The maximum line allocation achievable in this scenario

4.3 Bandwidth Over-allocation

Link Occupancy (%)

Effective Load

Real Load

Voice over IP: Overallocation Overhead (Pack Delay 32ms)
Offered Load (Erlang) 

<table>
<thead>
<tr>
<th></th>
<th>9.3</th>
<th>27.8</th>
<th>46.4</th>
<th>64.9</th>
<th>83.4</th>
<th>102.0</th>
<th>120.5</th>
<th>139.1</th>
<th>157.6</th>
<th>176.2</th>
<th>194.7</th>
</tr>
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</table>

Effective Link Load

Pack Delay 32 ms

Effective Load

20% 40% 60% 80%

Offered Load (Erlang) overheard (since the overhead and RTP headers have a constant length). On the other hand, a source and destination delay may be a larger issue.

Packet Delay

Voice over IP: Link Performance

4.2. THE PAYLOAD

between the effective and real loads. This section presents a qualitative

delay.

Voice over IP: Overallocation Overhead (Pack Delay 18ms)
The transport efficiency is considerably low. Even the lowest packetization delay allows for very, high real-time efficiency. In this case, a short packetization delay is not necessarily the best solution. In either case, a short packetization delay leads real-time traffic to the core link capacity. No extra effort to maintain the real-time traffic above the core link capacity is required for the transmission of real-time traffic. Thus, the real-time traffic will be transmitted even if a single packet is lost, which is acceptable. However, even if packet loss occurs, the network can still carry significant traffic. Figure 7 shows that a drop in packetization delay leads to significant improvements in the network. A 50% packetization delay with a packet-switching delay leads to significant improvements in the network. A 50% packetization delay improves the effective load, and a 100% packetization delay improves the effective load, too. Figure 6 shows that even a slight reduction in the packetization delay leads to significant improvements in the network. A 50% packetization delay improves the effective load, and a 100% packetization delay improves the effective load, too.

**Figure 6:** Blocking probability and effective load

**Figure 7:** Network efficiency: the effective load

**Figure 8:** Blocking probability and effective load

Network efficiency: the effective load.
In general, all the various packet technologies incur bandwidth overhead due to the various functions that must be performed on the packets. These functions include the following:

- **Routing**: Determining the route that the packet should take through the network.
- **Fragmentation**: Dividing a large packet into smaller packets that can be transmitted over the network.
- **Security**: Ensuring that the packet is transmitted securely.
- **QoS**: Ensuring that the packet is delivered on time.

These functions require additional processing and storage resources, which increase the overhead of the packet. The overhead is typically expressed as a percentage of the total data transmitted.

**Figure 1**: Impact of packetization delay over the real bandwidth of a phone call with various technologies.
The packet size can be expressed as a function of the packetization delay:

\[ P = \frac{1}{1 + \frac{\text{Packet Size}}{\text{Packet Delay}}} \]

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4.3 SONET/SDH and Voice Compression

The primary objective is to reduce the bit rate of voice and data to a lower level while maintaining the quality of the service. This is achieved through the use of voice compression and data compression techniques.

4.4 The Optimal IP Packet Size

Direct embedding a single phone call into a single IP packet is not feasible due to the large amount of overhead associated with IP packets. Instead, a more efficient approach is to use a group of phone calls to form a single packet. This approach is known as voice compression and is commonly used in voice-over-IP (VoIP) systems.

\[ \text{Packet Size} = \frac{\text{Packet Delay}}{1 + \frac{\text{Packet Size}}{\text{Packet Delay}}} \]

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5 Discussion

The number of hops traveled by a phone call increases as the number of nodes in the network increases. This leads to an increase in the total path length, which can negatively impact the overall performance of the network.

The equation for the effective load on the network is given by:

\[ \frac{D_{\text{Effective Load}}}{D_{\text{Total}}} \approx 1 + H \, D_{\text{Packet}} \]

where:
- \( D_{\text{Effective Load}} \) is the effective load on the network,
- \( D_{\text{Total}} \) is the total load on the network,
- \( H \) is the number of hops traveled by a call,
- \( D_{\text{Packet}} \) is the packet delay.

The optimal packet size depends on many parameters, including the effective load and the traffic density. The optimal packet size can be approximated using the following equation:

\[ D_{\text{Packet}} = \left( \frac{1}{H + 1} \right) \left( D_{\text{Total}} - D_{\text{Effective Load}} \right) \]

This equation is a good approximation when the effective load is low.

Because some of the delay components in (g) are inherent, some of the path delay is unavoidable. However, this delay can be minimized by optimizing the network parameters such as the packet size and the traffic density.

[Diagram showing the relationship between effective load and packet delay]


References

Acknowledgments

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

- The performance of the network is improved by packet switching, which reduces the latency and increases the throughput.
- The performance of the network is also improved by the use of controlled service network elements.
- The performance of the network is further improved by the use of peer-to-peer networking.

These results are summarized in the following table:

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