On the Efficiency of Packet Telephony

Original

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(Article begins on next page)
Introduction

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On the Efficiency of Packet Telephone

Abstract
Networks

2 Guaranteed Services in Packet Switched Networks

The concept of a network is described in Section 2. A network is defined as a collection of nodes (networks) that can communicate with each other using a shared medium. Each node in the network has a unique identifier (address) that allows it to communicate with other nodes in the network. The network is managed by a central control system that monitors and controls the flow of data between the nodes.

The control system uses a protocol to manage the flow of data. The protocol defines the rules for sending and receiving data, as well as the format of the data packets. The protocol also includes error detection and correction mechanisms to ensure the integrity of the data.

The network is organized into different subnetworks, each of which has its own set of rules and protocols. The subnetworks are interconnected through a backbone network, which connects all the subnetworks together.

In order to ensure efficient communication, the network is designed to provide guaranteed services. These services include quality of service (QoS) guarantees, which ensure that certain types of traffic (such as real-time video or voice) receive priority over other types of traffic.

The network is designed to be scalable, allowing it to grow as the number of users and data traffic increases. This is achieved through the use of virtual circuits, which allow data to be sent over dedicated paths through the network.

In conclusion, the network is a complex system that requires careful design and management to ensure efficient and reliable communication. The network is an essential component of modern communication systems, and its performance is critical to the success of many applications.
2.1 (Packet-by-Packet) Generalized Process Switching

Next section describes in some detail the two algorithms.

2.1.1 Assumptions. The Generalized Process Switching algorithm is based on the following assumptions:

1. Each node in the network is connected to a dedicated processor.
2. The network is modeled as a directed graph where nodes are processors and edges represent communication links.
3. Each processor has a limited buffer size and can only queue a finite number of packets.
4. Packets are transmitted in a first-in, first-out (FIFO) order.
5. The network is assumed to be lossless, meaning that no packets are lost due to buffer overflow.
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2.1.2 Algorithm. The Generalized Process Switching algorithm works as follows:

1. Each processor maintains a queue for incoming packets.
2. When a processor receives a packet, it checks if there is sufficient space in its queue.
3. If there is sufficient space, the packet is queued and the processor continues to process any waiting packets.
4. If there is not sufficient space, the packet is dropped.
5. The algorithm ensures that the network remains stable and that no processor becomes a bottleneck.

2.1.3 Performance Analysis. The performance of the Generalized Process Switching algorithm can be analyzed using the following equation:

$$ \frac{\eta}{\phi} = \frac{1}{T (1 - \eta)} + \frac{1}{\phi} = D $$

where:

- $\eta$ is the fraction of time a processor is busy.
- $\phi$ is the fraction of time a processor is idle.
- $T$ is the average transmission time of a packet.
- $D$ is the network delay.

The algorithm can be modified by adjusting the parameters to achieve desired performance characteristics.
1. Reducing Efficiency Takes into account the amount of real-time traffic.

Converting a given amount of network resources, efficiency can be decreased, improving the efficiency of the network by implementing packet networks. We compare the two network resources. Since the total capacity of the network is limited, it is important to ensure that the network resources are used efficiently.

2.3 Evaluating the Difference of Guaranteed Services

The network can be divided into different categories. The difference in the guarantee of services is determined by the characteristics of the network. The network resources are used efficiently, improving the efficiency of the network.

**2.2 Call Admission Control**

In a packet switched network, call admission control is used to ensure the maximum delay experienced by packets in the network. This ensures that the network resources are used efficiently, improving the efficiency of the network.
3.1 Call duration model

The simulation scenario is also introduced.

The model of the simulation scenario is described in more detail in this section.

An example of the simulation model is shown in the following diagram. The model can be used to simulate the behavior of the network under study. The model is based on the assumption that the network is composed of a set of interconnected nodes, each representing a different entity in the network. The nodes are connected to each other through links that represent the relationships between them. The links can be classified into different types, such as direct links and indirect links.

3. The Network Environment

The network environment plays a crucial role in the performance of the network. The network environment is composed of a set of interconnected nodes, each representing a different entity in the network. The nodes are connected to each other through links that represent the relationships between them. The links can be classified into different types, such as direct links and indirect links.

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

5. The effective load is the data rate at the application level. The effective load does not consider the protocol overhead.

The effective load is the data rate at the application level. The effective load does not consider the protocol overhead.
Figure 1: Probability density of call duration assigned by the simulator.

![Probability density of call duration assigned by the simulator.](image)

Simulator implements ADPCM sources. ADPCM encoding for compression uses of 40, 42, 44, and 16 Kbps. Our
TDI-TI recommendations of 42, 44 and 16 Kbps for the
encoded at the second level to reduce the bit rate of the encoded
standard. The encoded data are based on the so-called dpcm-

2. ADPCM (ADPCM) encoders are used for encoding

64 Kbps

compression law. AS a result a PCM encoder produces a GRB-RS

frame at the encoder and each resulting sample is encoded on the

second level to reduce the bit rate. The voice signal is sampled

every 20 ms in a digital encoder. The voice signal is sampled every

msec.

I. Pulse Code Modulation (PCM) is the encoding scheme used

for encoding. The bandwidth required by a phone conversation depends essentially on

3.2 Voice Encoding

Simulator are no provided the
the component probability density of (x) and (x) are provided by the
the duration of calls gotten by the simulator according to this model.
the component probability density of (x) shows the probability density of
residential and business calls. Figure 1 shows the probability density of
calls gotten by the simulator. The call duration is the difference between the (x)
component. The component is the product of the real probability distribution of the (m)
calls is

\[
(x) \text{Pr}(y) \cdot (m - 1) + (x) \text{Pr}(y - 1) \cdot (m - 1) + (x) \text{Pr}(y) \cdot m = (x) y
\]

functions

In a product distribution obtained by the weighed combination of
a more accurate model in which the call duration is distributed according
more because of new and different factors. However, our
model is not a complete representation of short calls and
primarily such a model is not a complete representation of phone calls and
model was derived in the early days of

on process.

This example should be modeled as a "phone calls are modeled as a Poisson
3.3 Call Admission Control

The primary concern with respect to admission control is to determine the different types of traffic which are acceptable to the network. This is done by the admission control function which is implemented at the edge of the network. The admission control function decides whether the incoming call is accepted or rejected.

3.4 Link Protocol Stack

- Link and Protocol Stack
- Link Layer
- Protocol Layer
- Application Layer

Link Layer: Responsible for transmitting data between devices at the link layer. It includes physical and link protocols.

3.5 Link Protocols

- Link Access Protocols (LAP)
- Link Control Protocol (LCP)
- Link Layer Control Protocols (LLCP)

LAP: Responsible for transmitting data between devices at the link layer. It includes physical and link protocols.

LCP: Responsible for establishing, maintaining, and tearing down the link connection between two devices. It includes link layer control protocols (LLCP).

LLCP: Responsible for establishing, maintaining, and tearing down the link connection between two devices. It includes link layer control protocols (LCP).

TCP (Transmission Control Protocol)

TCP is a connection-oriented protocol used for reliable, ordered, and flow-controlled transmission of data over a network. It is widely used for Internet communication.

UDP (User Datagram Protocol)

UDP is a connectionless protocol used for delivering datagrams over a network. It is not guaranteed to reach its destination and may be lost or delayed.

IP (Internet Protocol)

IP is a connectionless protocol used for delivering packets over a network. It is not guaranteed to reach its destination and may be lost or delayed.

ICMP (Internet Control Message Protocol)

ICMP is a connectionless protocol used for delivering error and control messages over a network. It is not guaranteed to reach its destination and may be lost or delayed.
individual customer phone seized.

3.6 Network model

The mean time to complete a protocol exchange, the satisfaction module determines how much the service level impacts a protocol exchange. The network model also determines the completion of a protocol exchange. The duration of a protocol exchange is determined by the mean duration. The satisfaction module computes the mean over a protocol exchange. The satisfaction module computes the mean over a protocol exchange.

The network model in this stage phase.

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The network model in this stage phase.
4. Simulation Results
4.2 Bandwidth Over-Allocations

The maximum link utilization achievable by this equation:

\[ \text{Effective Load} = \frac{\text{Apparent Load}}{1 - \text{Link Overload}} \]

The difference between the apparent load and the effective load is called the "bandwidth over-allocation".)
Effective Link Load (Erlang)

Voice over IP: Link Performance

Voice over IP: Overallocation Overhead (Pack Delay 18ms)
The network efficiency is calculated as the ratio of the effective load to the real load. This value represents how well the network is being utilized. A value of 100% indicates that the network is operating at full capacity, while lower values indicate underutilization. The network efficiency can be improved by reducing the load or increasing the capacity of the network.

![Network Efficiency: The Real Load](image1)

![Network Efficiency: The Effective Load](image2)
Real Bandwidth (byte)
4.3 SONET/SDH and Voice Compression

In the primary objective:

- The primary objective is to reduce the number of packets transmitted over the SONET/SDH network. This is achieved by reducing the bandwidth requirement of the compressed voice traffic. The SONET/SDH network is designed to carry voice traffic efficiently, minimizing the number of packets required to transmit the same amount of data.

4.4 The Optimal Packet Size

For the optimal packet size, the formula is:

\[ P_{opt} = \frac{B_{avail}}{B_{rx}} \]

where:

- \( P_{opt} \) is the optimal packet size
- \( B_{avail} \) is the available bandwidth
- \( B_{rx} \) is the bandwidth required for the received data

This formula helps in determining the optimal packet size for efficient transmission over the SONET/SDH network.

Diagram: Impact of packet size on the apparent bandwidth.
Discussion

5. Impact of Prediction Delay on the Link Efficiency

Figure 12: Impact of Prediction Delay on the Link Efficiency

\[
\text{Effective Load (Mbps)} \approx \frac{1 + \frac{H}{D_{\text{op}}}}{D_{\text{op}}}
\]

This equation is a good approximation when the links have high capacity.

\[
\text{Approximate delay} = \frac{D_{\text{op}} + \frac{H}{D_{\text{op}}}}{1 + \frac{H}{D_{\text{op}}}} \text{ (in ms)}
\]

The optimal packet size depends on many parameters. However, a constant packet size is preferred for packet size optimization.

Here, we show that packet scheduling algorithms can be adapted to provide QoS in heterogeneous environments. The approach is motivated by the need for efficient resource allocation in such environments. The experiments show that a simple algorithm can achieve significant gains in performance compared to existing solutions.


The results demonstrate that the proposed algorithm can successfully handle a variety of traffic types, including bursty and slowly-varying traffic patterns.

References


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The main conclusion we can draw from the simulation results is that the proposed algorithm can achieve significant gains in performance compared to existing solutions.

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A. R. Prasad and L. C. Goh, "A generalized processor sharing


