Introduction

Abstract

January 6, 1999

101 29 Town, ITALY
C/o Onda Dahr, Amerazza 24
Pdolomo di Teramo
Department of Automation e Informatica
Mario Baldu, Davide Bertaini and Punio Russo

On the Efficiency of Packet Telephony
Networks

2 Guaranteed Services in Packet Switched Networks

The primary objective of the design of a network intended for a given class of packet switches is to provide a guaranteed service level for that class, even if it is provided as a logical service. In the IP-based networks, the service is provided as a logical service, and the service level is guaranteed by the network provider. In this section, we discuss the mechanisms used to provide the required service level.

The paper is structured as follows. Section 2 discusses the network model, the service level, and the network resource requirements. Section 3 describes the mechanisms used to provide the required service level. Section 4 analyzes the performance of the network, and the section concludes with a discussion of the implications of the results.
This bound can be intuitively explained by considering that a burst of packets, \( D = \frac{\rho \eta}{1 - \eta} \), should be delivered within the channel transmit time. Hence, it is clear that the channel must be capable of accommodating \( \rho \eta \) packets simultaneously. However, this bound is not necessarily tight because it does not account for the overhead in the network.

Therefore, the tightness of this bound is not established in general. However, in the special case where the channel is known to be capable of accommodating \( \rho \eta \) packets simultaneously, the bound becomes tight.

Theorem 2.1 (Packet-by-packet Generalized Process-Smoothing)

Next section describes in some detail the two algorithms that will be used to compute the two processes, the channel capacity and the average delay. The algorithms are based on the fact that the channel capacity is equal to the maximum of the two values:

1. The capacity of the channel, which is the maximum number of packets that can be transmitted in a given time.
2. The average delay of the packets, which is the average time it takes for a packet to be transmitted.

The algorithms are based on the fact that the channel capacity is equal to the maximum of the two values:

- The capacity of the channel, which is the maximum number of packets that can be transmitted in a given time.
- The average delay of the packets, which is the average time it takes for a packet to be transmitted.

The algorithms are based on the fact that the channel capacity is equal to the maximum of the two values:

- The capacity of the channel, which is the maximum number of packets that can be transmitted in a given time.
- The average delay of the packets, which is the average time it takes for a packet to be transmitted.
2.3 Evaluating the Efficiency of Guaranteed Services

Over Packet Networks

I. Reducing Efficiency losses into account can the amount of real-time traffic

A contrast of the amount of traffic and reserved capacity can be realized by

Converting the amount of traffic and reserved capacity into the efficiency of

the network. This can be done by comparing the amount of traffic with

reserved capacity. This is achieved by calculating the ratio of traffic to

reserved capacity. This ratio can then be used to evaluate the efficiency of

the network.

The CVC (Constant Varnish Capacity) can be calculated by the formula:

CVC = Traffic / Reserve Capacity

This formula allows for the calculation of the CVC, which can then be used to

evaluate the efficiency of the network.

II. Meeting the Demand for Guaranteed Services

The CVC is used to determine the amount of capacity that is reserved for

guaranteed services. This can be done by calculating the CVC and then using

this value to reserve the necessary capacity.

For example, if the CVC is 0.8, this means that 80% of the reserved capacity is

utilized. This can be used to ensure that the necessary capacity is reserved for

guaranteed services.

The CVC can also be used to determine the amount of capacity that is reserved

for non-guaranteed services. This can be done by subtracting the CVC from 1.0

and then using this value to reserve the remaining capacity.

For example, if the CVC is 0.8, this means that 20% of the reserved capacity is

utilized for non-guaranteed services. This can be used to ensure that the

necessary capacity is reserved for non-guaranteed services.

The CVC can be used to determine the amount of capacity that is reserved for

both guaranteed and non-guaranteed services. This can be done by adding the

CVC to 1.0 and then using this value to reserve the necessary capacity.

For example, if the CVC is 0.8, this means that 120% of the reserved capacity is

utilized. This can be used to ensure that the necessary capacity is reserved for

both guaranteed and non-guaranteed services.

The CVC can also be used to determine the amount of capacity that is reserved

for non-guaranteed services. This can be done by subtracting the CVC from 1.0

and then using this value to reserve the remaining capacity for non-guaranteed

services.

For example, if the CVC is 0.8, this means that 20% of the reserved capacity is

utilized for non-guaranteed services. This can be used to ensure that the

necessary capacity is reserved for non-guaranteed services.

The CVC can be used to determine the amount of capacity that is reserved for

both guaranteed and non-guaranteed services. This can be done by adding the

CVC to 1.0 and then using this value to reserve the necessary capacity for both

guaranteed and non-guaranteed services.

For example, if the CVC is 0.8, this means that 120% of the reserved capacity is

utilized for both guaranteed and non-guaranteed services. This can be used to

ensure that the necessary capacity is reserved for both guaranteed and

non-guaranteed services.
3.1 Call Duration Model

The simulation scenario is also introduced.

The rest of the section describes in more detail the simulator models.

The objective of the simulation is to test some of the network models. Each model captures the behavior of each cell and generates both the call handling and the call dropping. The models are designed to capture the behavior of the network in a way that allows for the evaluation of the performance of the network under different conditions. The models also allow for the evaluation of the performance of the network under different traffic conditions.

3. The Simulation Environment

The performance of the network can be affected by many factors, such as the location of the user, the network type, and the network technology. The performance of the network can be improved by improving the performance of the network elements. The performance of the network can be improved by improving the performance of the network elements. The performance of the network can be improved by improving the performance of the network elements.
3.2 Voice Encoding

Simulators are the same. Probability density

\[
\frac{1}{2\pi} |\mathbf{f}(x)\mathbf{a}| \cdot |\mathbf{f}(x)\mathbf{b}| \cdot |\mathbf{f}(x)\mathbf{c}| \cdot m \cdot (f(x))
\]

These are exponentially distributed (i.e., phone calls are modeled as Poisson processes).
3.4 Call Admission Control

The purpose of the simulation study is to assess the efficiency provided by the different admission control algorithms. In order to determine the actual performance of the algorithms, we compare the simulation results with the theoretical analysis. The theoretical analysis is based on the assumption that the network is perfectly balanced, i.e., the link utilization is equal for all links. In order to verify the assumptions, we simulated a network with varying link utilization and compared the simulation results with the theoretical analysis.

The results show that the admission control algorithms perform well under varying network conditions. The algorithms are able to balance the link utilization and ensure a fair distribution of traffic. However, the algorithms do not perform well under highly unbalanced network conditions. In such cases, the algorithms may need to be modified to improve their performance.

3.5 Link model and Protocol Stack

A protocol stack is a layered model that describes the communication protocol used by a computer network. The protocol stack is divided into multiple layers, each of which is responsible for a specific function. The layers are connected by interfaces, which define the interface between two layers.

The protocol stack is often referred to as a stack because the layers are stacked on top of each other. The layers are organized in a hierarchical manner, with lower layers providing services to higher layers. The layers are organized as follows:

1. Application layer: This layer provides services to end-users, such as file transfer, email, and web browsing.
2. Transport layer: This layer provides end-to-end communication services, such as reliable data transfer and flow control.
3. Network layer: This layer provides services for routing and network addressing, such as IP addressing and routing.
4. Link layer: This layer provides services for data transmission over a physical medium, such as Ethernet and wireless networks.
5. Physical layer: This layer provides services for physical transmission of data over a medium, such as fiber optics and coaxial cables.

The protocol stack is often used to describe the communication protocols used by a computer network. The layers are organized in a hierarchical manner, with lower layers providing services to higher layers. The layers are organized as follows:

1. Application layer: This layer provides services to end-users, such as file transfer, email, and web browsing.
2. Transport layer: This layer provides end-to-end communication services, such as reliable data transfer and flow control.
3. Network layer: This layer provides services for routing and network addressing, such as IP addressing and routing.
4. Link layer: This layer provides services for data transmission over a physical medium, such as Ethernet and wireless networks.
5. Physical layer: This layer provides services for physical transmission of data over a medium, such as fiber optics and coaxial cables.

The protocol stack is often used to describe the communication protocols used by a computer network. The layers are organized in a hierarchical manner, with lower layers providing services to higher layers. The layers are organized as follows:

1. Application layer: This layer provides services to end-users, such as file transfer, email, and web browsing.
2. Transport layer: This layer provides end-to-end communication services, such as reliable data transfer and flow control.
3. Network layer: This layer provides services for routing and network addressing, such as IP addressing and routing.
4. Link layer: This layer provides services for data transmission over a physical medium, such as Ethernet and wireless networks.
5. Physical layer: This layer provides services for physical transmission of data over a medium, such as fiber optics and coaxial cables.

The protocol stack is often used to describe the communication protocols used by a computer network. The layers are organized in a hierarchical manner, with lower layers providing services to higher layers. The layers are organized as follows:

1. Application layer: This layer provides services to end-users, such as file transfer, email, and web browsing.
2. Transport layer: This layer provides end-to-end communication services, such as reliable data transfer and flow control.
3. Network layer: This layer provides services for routing and network addressing, such as IP addressing and routing.
4. Link layer: This layer provides services for data transmission over a physical medium, such as Ethernet and wireless networks.
5. Physical layer: This layer provides services for physical transmission of data over a medium, such as fiber optics and coaxial cables.

The protocol stack is often used to describe the communication protocols used by a computer network. The layers are organized in a hierarchical manner, with lower layers providing services to higher layers. The layers are organized as follows:

1. Application layer: This layer provides services to end-users, such as file transfer, email, and web browsing.
2. Transport layer: This layer provides end-to-end communication services, such as reliable data transfer and flow control.
3. Network layer: This layer provides services for routing and network addressing, such as IP addressing and routing.
4. Link layer: This layer provides services for data transmission over a physical medium, such as Ethernet and wireless networks.
5. Physical layer: This layer provides services for physical transmission of data over a medium, such as fiber optics and coaxial cables.

The protocol stack is often used to describe the communication protocols used by a computer network. The layers are organized in a hierarchical manner, with lower layers providing services to higher layers. The layers are organized as follows:

1. Application layer: This layer provides services to end-users, such as file transfer, email, and web browsing.
2. Transport layer: This layer provides end-to-end communication services, such as reliable data transfer and flow control.
3. Network layer: This layer provides services for routing and network addressing, such as IP addressing and routing.
4. Link layer: This layer provides services for data transmission over a physical medium, such as Ethernet and wireless networks.
5. Physical layer: This layer provides services for physical transmission of data over a medium, such as fiber optics and coaxial cables.
3.6 Network model

The network model is constructed using a combination of statistical and analytical methods. The network model is designed to simulate the behavior of the network under various conditions. The network model is composed of three main components: the statistical model, the analytical model, and the simulation model. The statistical model is used to calculate the probability distribution of the network traffic. The analytical model is used to calculate the performance metrics of the network. The simulation model is used to simulate the behavior of the network under different conditions.

The performance of the network is evaluated using a set of metrics. These metrics include the network throughput, the network latency, and the network delay. The network throughput is calculated by dividing the total number of packets sent by the total number of packets received. The network latency is calculated by dividing the total number of packets sent by the total number of packets received. The network delay is calculated by dividing the total number of packets sent by the total number of packets received.
4 Simulation Results

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

Figure 4: Network Topology used in the simulation.
4.1 Bandwidth Over-allocation

![Graph showing efficiency index on link T0 vs. Link P0 with high packetization.]

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

The maximum utilization achieved in this scenario

With links to a packet switched network.

The difference between the apparent load and the effective load (\(L_0\)) is the effective load (\(L_{eff}\)). The difference between the apparent load and the effective load (\(L_0\)) is the apparent load (\(L_{app}\)). The difference between the apparent load and the effective load (\(L_0\)) is the effective load (\(L_{eff}\)).
Offered Load (Erlang)

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

Effective Link Load (Erlang)

<table>
<thead>
<tr>
<th>Effective Link Load (Erlang)</th>
<th>20%</th>
<th>40%</th>
<th>60%</th>
<th>80%</th>
</tr>
</thead>
</table>

Effective Load

<table>
<thead>
<tr>
<th>Effective Load</th>
<th>200</th>
<th>300</th>
<th>400</th>
<th>500</th>
</tr>
</thead>
</table>

Apparent Load

<table>
<thead>
<tr>
<th>Apparent Load</th>
<th>600</th>
<th>700</th>
<th>800</th>
<th>900</th>
</tr>
</thead>
</table>

Circuit Switching

Real Load

Pack Delay 32 ms

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

Pack Delay 18 ms

20% 40% 60% 80%

Link Occupancy (%)

Circuit Switching (PCM64)

Circuit Switching

Voice over IP

4.2.1 The Payload

utilization efficiency

assumption of the impact of the two parameters on the real-time network between the effective and real loads. This section presents a qualitative

delay

Payload Efficiency

Voice over IP: Overload (Pack Delay 1 ms)

Voice over IP: Overload (Pack Delay 1 ms)

Offered Load

since the PPP/Pcap/IP/RTP packets have a constant length.

is higher in the former case.

However, there is another issue to consider: namely the amount of data.

is also depends on the effective link load. A higher link load is expected to provide a higher efficiency. However, a higher packet load is expected to provide a higher efficiency. However, a

this can be noted by observing that in Figure 6 the difference between the

Figure 6: Impact of packet size over the efficiency: effective load.
The transport efficiency is considerably low. For the D5 and D6 transport channels, a short protection delay is not necessary in the protection. However, more protection delay leads to increased protection of real-time traffic and brings other problems. If the protection delay leads to increased protection of real-time traffic, then the real-time traffic over the network is interrupted. In contrast, protection delay leads to increased protection of real-time traffic and brings other problems. If the protection delay leads to increased protection of real-time traffic, then the real-time traffic over the network is interrupted.

Figure 6: Blocking probability and real load.

![Figure 6: Blocking probability and real load.](image)

Figure 7: Network efficiency: the real load.

![Figure 7: Network efficiency: the real load.](image)
Packet Delay (ms)

- 26.5
- 25.0
- 22.0
- 19.0
- 17.5
- 16.0
- 11.5
- 10.0
- 8.5
- 7.0

Circuit Switching (ADPCM32)

Real Bandwidth (byte)

- 60000
- 80000

Voice over IP over SONET

ATM

Voice over ATM

Packetization Overhead: the Real Bandwidth of a Phone

The header size depends on the protocol architecture employed in the network. Different network architectures have different header sizes. In ATM, the header consists of a cell header (64 bytes) plus overheads (IP, PVC) for each IP packet. In SONET, the header includes the overheads (SONET/SDH) for each IP packet. ATM and SONET provide different levels of service, with ATM being more flexible and SONET being more rigid. The header size is determined by the specific protocol used and the network configuration.
The optimal IP packet size is given by:

\[ I = \frac{1}{2} \cdot \frac{\text{Packet Size} \cdot \text{Packet Loss}}{1 + \text{Packet Size} \cdot \text{Packet Loss}} \]

The packet size can be expressed as a function of the packetization delay:

\[ I = \frac{1}{2} \cdot \frac{\text{Packet Size} \cdot \text{Packet Loss}}{1 + \text{Packet Size} \cdot \text{Packet Loss}} \]

The primary objective of the packetization delay is to ensure that the data received is exactly the same as the data sent. In other words, the delay in the network must be minimal to maintain the integrity of the data. This section analyzes the delay bound formulas used to define the optimal packet size.

**4.3 SONET/SDH and Voice Compression**

The addition of voice compression to the packetization delay over the application bandwidth of a phone call allows a single phone call to be decompressed and then reconnected to the network, thereby maintaining the integrity of the voice data. This process is crucial in ensuring that the voice data is transmitted accurately and efficiently.
5 Discussion

Figure 12: Impact of Reservation Delay on the Link Efficiency

\[ \frac{1 + H}{D_{lookout}} \approx \frac{1}{D_{look}} \]

Since some of the delay components in (6) vanish when formula (4) and (8) are applied, the optimal packet size depends on many parameters. However, a compact formula for the optimal packet size is given by:

\[ \frac{1 + H}{n_{prop}} \frac{D_{prop}}{D_{lookout}} - D_{prop} = D_{lookout} \]