Introduction

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Abstract

On the Efficiency of Packet Telephony

with maximum real-time echo is desired multiplying the packet size

which minimize real-time echo is desired multiplying the packet size

Results show that packet size possibly constraining the real-time echo

calls are the simulation optimal

minimize delay bound for each call. Simulation data are accepted and packet

According to the multiplicity of the resources inserted to produce a video

We developed a real-time simulation with allows a general video

The work presents a study on the efficiency of packet switching and er-

information of real-time and non real-time services.

lower cost and higher manageability than circuit switching.

The informaion of a mcomeo switching network because it includes

This paper presents a study on the efficiency of packet switching in pro-

1
2 Guaranteed Services in Packet Switched Networks

In Section 2, Guaranteed Services in Packet Switched Networks, we will discuss the implementation of guaranteed services in packet-switched networks. The focus will be on the design and operation of a network that provides differentiated services, such as QoS (Quality of Service) guarantees, to different types of traffic. The paper will discuss the architecture of a network designed to carry the different classes of traffic, focusing on the protocol stack used to support these services. The goal is to provide a framework for implementing differentiated services in a packet-switched network, allowing for the provision of quality guarantees to different types of traffic.
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1. Performance Equivalency

2. The formula to determine the amount of real-time impact over Packet Networks

2.2 Call Admission Control

2.4 Implementing the Admission Control

2.5 Summary

3. Conclusion

4. References
The simulation scenario is also introduced.

The rest of the section describes in more detail the simulation scenario.

The model is described by a set of parameters that are important from simulation and analytical points of view.

We define a set of four efficiency parameters that are important from simulation and analytical points of view.

The efficiency of the network is increased by a factor of the order of 10. The efficiency of the network is also increased by a factor of the order of 10.
1. Pulse Code Modulation (PCM) is the encoding scheme used in digital telephone networks. The voice signal is sampled exactly 8 times per second and each resulting sample is encoded on a byte using a non-linear encoding technique. A PCM encoder takes a sample of the voice signal and converts it into a series of binary numbers. These numbers represent the amplitude of the voice signal at that moment. The process is reversed at the decoder, where the binary numbers are converted back into an analog signal that can be played through a speaker or sent over a telephone line.

2. ADPCM (Adaptive Differential Pulse Code Modulation) is a modified form of PCM that is used in digital telephone networks to reduce the amount of data that needs to be transmitted. ADPCM works by predicting the next sample of the voice signal based on the previous samples and then encoding the difference between the predicted value and the actual value. This reduces the amount of data that needs to be transmitted because the difference between the predicted value and the actual value is much smaller than the voice signal itself.

3. Voice Coding

Voice coding is the process of converting the analog voice signal into a digital format that can be transmitted over a data network. There are several different voice coding techniques, each with its own advantages and disadvantages. The most common voice coding technique is Pulse Code Modulation (PCM), which is used in digital telephone networks.

Voice coding is an important part of telecommunication systems, as it allows voice signals to be transmitted over data networks. Without voice coding, it would be impossible to transmit voice signals over data networks, as the analog voice signal cannot be transmitted directly over a data network.

Voice coding techniques are constantly evolving, as new technologies are developed to improve the quality of voice transmission. New voice coding techniques are being developed that can transmit voice signals over data networks with even higher quality and lower latency than current voice coding techniques.
**3.3 Link model and Protocol Stack**

A protocol or link layer (also called link layer protocol) can be defined as a set of rules by which two network nodes communicate over a particular link. The link layer is typically responsible for providing a reliable connection between two devices, ensuring that data is transmitted accurately and efficiently. This layer is crucial for the smooth operation of the entire communication process.

**3.3.1 Link Layer Protocol**

The link layer is responsible for providing a reliable connection between two devices over a link. It establishes, maintains, and terminates the connection between the devices. The link layer protocol is based on the concept of a link, which is a connection between two devices that provides a means for exchanging data. The link layer protocol is responsible for ensuring that data is transmitted accurately and efficiently over the link.

**3.3.2 Link Layer Admissions Control**

The link layer is responsible for accepting new connections onto the link while maintaining the quality of existing connections. The link layer must be able to accept or reject new connections based on various factors such as the available bandwidth, the congestion level, and the QoS requirements of the new connection.

**3.3.3 Link Layer Error Control**

The link layer is responsible for detecting and correcting errors that occur during data transmission. Errors can occur due to various reasons such as noise, interference, or hardware failures. The link layer must be able to detect and correct these errors to ensure the accuracy of the transmitted data.

**3.3.4 Link Layer Flow Control**

The link layer is responsible for controlling the flow of data to prevent congestion and ensure that the network resources are used efficiently. The link layer must be able to control the rate at which data is transmitted to prevent the network from becoming congested.

**3.3.5 Link Layer Addressing**

The link layer is responsible for assigning unique addresses to devices on the network. These addresses are used to identify the devices and provide a means for transmitting data to a specific device.

**3.3.6 Link Layer Security**

The link layer is responsible for providing security to the data transmitted over the link. This includes encrypting the data to prevent unauthorized access and ensuring the integrity of the data.

**3.3.7 Link Layer Management**

The link layer is responsible for managing the link and ensuring its proper operation. This includes monitoring the link quality, detecting failures, and configuring the link as needed.

**3.3.8 Link Layer Monitoring**

The link layer is responsible for monitoring the link to detect and respond to failures. This includes monitoring the link quality, detecting failures, and configuring the link as needed.

**3.3.9 Link Layer Access Control**

The link layer is responsible for controlling access to the link. This includes granting or denying access to devices based on various factors such as the user's identity, the device's authentication credentials, and the network's security policies.

**3.3.10 Link Layer Configuration**

The link layer is responsible for configuring the link to meet the specific requirements of the network. This includes configuring the link parameters, setting up the link connections, and configuring the link's security settings.

**3.3.11 Link Layer Troubleshooting**

The link layer is responsible for troubleshooting any issues that arise on the link. This includes detecting and diagnosing problems, identifying the root cause of the issue, and implementing a solution to resolve the problem.

**3.3.12 Link Layer Performance**

The link layer is responsible for ensuring the performance of the link. This includes monitoring the link's performance, identifying performance bottlenecks, and implementing solutions to improve the link's performance.

**3.3.13 Link LayerQoS**

The link layer is responsible for ensuring that the QoS requirements of the network are met. This includes allocating resources to different network flows, prioritizing traffic, and adjusting the link's settings to meet the QoS requirements.

**3.3.14 Link Layer Optimization**

The link layer is responsible for optimizing the link to improve its performance. This includes implementing techniques such as flow control, congestion avoidance, and error correction to enhance the link's performance.

**3.4 Link Layer Protocol**

The link layer protocol is responsible for controlling the flow of data over the link. It establishes, maintains, and terminates the connection between the devices. The link layer protocol is based on the concept of a link, which is a connection between two devices that provides a means for exchanging data. The link layer protocol is responsible for ensuring that data is transmitted accurately and efficiently over the link.
individual customers' phone sizes.

Table 2. Protocol sizes used in the simulations.
4 Simulation Results

The network topology used in the simulations (see Figure 4) has been modeled after the actual Telecom Italia’s telephone network in order to reproduce a quite realistic test environment.

Local and toll offices have been replaced by routers whereas local exchanges have been upgraded with the packetization functionality. The physical length and capacity of links are the same as the real phone network.

The typical domestic long distance call over the Telecom Italia’s network crosses at most two local offices and two toll offices. In the scenario depicted in Figure 4, two long distance calls are originated from two different areas (local offices $LO_1$ and $LO_3$) toward the same area ($LO_2$).

Since the packetization process is carried out by local exchanges, from the simulation standpoint they can be assumed as the endpoints of phone calls. These are originated from each local exchange connected to $LO_1$ and $LO_3$ and are directed towards every local exchange connected to $LO_2$. Unless specified differently, the ADPCM32 coding scheme is exploited.

Simulations have been run with an increasing offered load in terms of calls per hour in order to determine the maximum achievable utilization of link $TO_2 - LO_2$. Since the actual offered load depends both on the calls duration and frequency, in the rest of the paper it is expressed using a measurement unit known as $Erlang$. The Erlang, i.e., call frequency times average call duration, is the typical measurement unit used in telephony to quantity the offered load.

As explained in Section 3.4, the bound on the delay introduced by the network on the packets belonging to a flow depends on the apparent load of that flow. Given the maximum end-to-end delay constraint of 100 ms, the amount of bandwidth that must be reserved to a flow, called the apparent bandwidth, can happen to be larger than the minimum amount of bandwidth required to actually transmit the data, referred to as the real load. The rest of this section is devoted to identifying the factors that affect the efficiency of packet switching and to determine the trade-off between real-time efficiency and traffic efficiency, whenever possible. Section 4.1 analyzes the difference between apparent and real load on the network and highlight the consequences on network utilization efficiency. Section 4.2 studies the impact of packetization on bandwidth allocation, whereas Section 4.3 discusses the limitation of the current circuit switching technology for transporting compressed voice. Lastly, Section 4.4 explains how to determine the packet size which maximizes real-time efficiency.
4.2 Packet Over-allocation

Compared to the previous section, this section introduces a model to evaluate the impact of packet over-allocation. In previous work, the packet overhead was calculated by the difference between the Link Occupancy (LO) of the network and the effective load of the network. However, in this section, the model is extended to include the difference between the Link Occupancy (LO) of the network and the effective load of the network. This allows for a more accurate evaluation of the packet overhead. The model is based on the difference between the Link Occupancy (LO) of the network and the effective load of the network. The model is defined as:

\[ \text{Packet Over-allocation} = \text{Link Occupancy} - \text{Effective Load} \]

4.3 Bandwidth Over-allocation

If there is an additional load on the network, the bandwidth over-allocation can be calculated using the following formula:

\[ \text{Bandwidth Over-allocation} = \text{Link Occupancy} - \text{Real Load} \]

Where:
- Link Occupancy (LO) is the percentage of the link capacity that is currently being used.
- Effective Load (EL) is the actual load on the link, excluding any overhead.
- Real Load (RL) is the actual load on the link, including any overhead.

The diagram below illustrates the relationship between the Link Occupancy, Effective Load, and Real Load, and how they contribute to the Bandwidth Over-allocation.
4.21 THE PAYLOAD

### Effective Load

![Effective Load vs. Overload Load](image)

- **Voice over IP: Link Performance**
- **Circuit Switching (PCM 64)**
- **Circuit Switching**
- **ADPCM 32**

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**Effective Load**

- **Overload Load**
- **Circuit Switching**
- **ADPCM 32**

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

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**Effective Load**

- **Overload Load**
- **Circuit Switching**
- **ADPCM 32**
The transport efficiency is considerably low. When the link is in use, the rate of packetization delay is not necessary the best solution. In this case a short packetization delay is also necessary. When a packet is transmitted over a network of real-time traffic, the network efficiency is also important. In this case a short packetization delay is not necessary. In this case a short packetization delay is also necessary. When a packet is transmitted over a network of real-time traffic, the network efficiency is also important. In this case a short packetization delay is also necessary. When a packet is transmitted over a network of real-time traffic, the network efficiency is also important. In this case a short packetization delay is also necessary. When a packet is transmitted over a network of real-time traffic, the network efficiency is also important. In this case a short packetization delay is also necessary. When a packet is transmitted over a network of real-time traffic, the network efficiency is also important. In this case a short packetization delay is also necessary. When a packet is transmitted over a network of real-time traffic, the network efficiency is also important. In this case a short packetization delay is also necessary.
Circuit Switching (ADPCM32)
The packet size can be expressed as a function of the packetization delay:

$$\frac{p}{1} = \frac{q + \text{Packet Size} \cdot D}{1}$$

The optimal packet size is obtained when the delay is zero. The number of packets is calculated by subtracting the number of packets in a single channel from the delay of the packetized channel. When the delay is zero, the number of packets in a single channel is obtained by subtracting the delay of the packetized channel from the delay of the packetized channel. The number of packets in a single channel is obtained by subtracting the delay of the packetized channel from the delay of the packetized channel. The number of packets in a single channel is obtained by subtracting the delay of the packetized channel from the delay of the packetized channel. The number of packets in a single channel is obtained by subtracting the delay of the packetized channel from the delay of the packetized channel.
**Discussion**

Figure 12: Impact of Prediction Accuracy on the Pack Efficiency

$$
\text{Effective Load (ms)} = \frac{I + H}{D_{\text{pack}}} 
$$

Optimal Pack Delay

$$
D_{\text{pack}} = \frac{I + H}{D_{\text{opt}}} 
$$

The number of hop inversions by a phone call is a key factor in determining the number of optimal packets. However, the number of optimal packets also depends on many parameters. Hence, it is important to optimize the optimal packet size. In our network, we have implemented a technique that allows for the efficient handling of delay. The technique is based on the principle of optimizing the optimal packet size to best fit the network load requirements.

The call admission scheme is based on the proposed QoS model. The scheme is designed to support a range of QoS requirements, including real-time and non-real-time applications. The scheme is designed to be flexible and scalable, allowing for the addition of new services and the modification of existing services.

The main contributions of the scheme are:

- The scheme provides a flexible and scalable approach to call admission control.
- The scheme allows for the provisioning of multiple QoS classes, each with different service guarantees.
- The scheme is designed to be adaptable to different network environments.
- The scheme is designed to be robust against changes in network conditions.

Figure 1: Illustration of the call admission scheme.

References

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