On the Efficiency of Packet Telephony

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1 Introduction

Abstract

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On the Efficiency of Packet Telephony
Networks

2 Guaranteed Services in Packet Switched Networks

Research is ongoing in Section 2. The results shown in Section 4. Combined Forwarding and Priority of
the Internet are achieved through the use of optical fiber to provide a separate, high-speed network for
high-priority traffic, such as voice and video. Forwarding and Priority are achieved through the
use of specialized devices, such as routers and switches, that prioritize traffic based on
predefined rules. The result is a network that can support multiple services, including
high-speed Internet access, voice over IP (VoIP), and video conferencing.

The paper is structured as follows:

1. Introduction
2. Related Work
3. Architecture
4. Protocols
5. Performance
6. Conclusion

The introduction provides an overview of the research topic, including its importance and relevance to
current trends in networking and communication. The related work section discusses previous
research in the field and highlights the unique contributions of the proposed solution.

The architecture section describes the overall design of the network, including the hardware and
software components. The protocols section details the specific protocols used to manage
traffic and ensure reliable communication. The performance section evaluates the
network's performance in terms of latency, bandwidth, and other key metrics.

The conclusion summarizes the key findings and highlights the potential impact of the
research on the field of networking. The paper concludes with a discussion of
future work and opportunities for further research.

The B-ISDN and related research efforts are now underway to enhance the
performance and capabilities of the network infrastructure.
...
1. Real-time efficiency allows the accurate amount of real-time traffic.

2. Over Packet Networks

2.3 Evaluating the Efficiency of Guaranteed Services

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The CC is a protocol that can coordinate different types of traffic and their service requirements. It is based on the concept of a Guaranteed Service (GS) layer, which ensures a fixed amount of bandwidth for a set of connections. The GS layer is responsible for controlling the amount of traffic that can be sent on the network at any given time, ensuring that all connections receive the guaranteed bandwidth. This is achieved through the use of admission control mechanisms, which determine whether a new connection can be established given the current load on the network.

The CC protocol also includes mechanisms for ensuring fairness among connections. When a new connection requests bandwidth, the CC checks whether the current load on the network will exceed the available resources. If the load is expected to exceed the available resources, the CC will deny the request and notify the source node of the denial. This ensures that the network remains stable and that all connections receive the guaranteed bandwidth.

In a packet-switched network, where connections are established and terminated dynamically, the CC is responsible for maintaining balance between different types of traffic. The CC ensures that the network resources are allocated fairly among all connections, preventing any single connection from monopolizing the network resources. This is achieved through the use of admission control mechanisms, which ensure that the network remains balanced and that all connections receive the guaranteed bandwidth.

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2. Call Admission Control

The CC protocol includes mechanisms for call admission control, which determine whether a new call can be established given the current load on the network. These mechanisms ensure that the network remains balanced and that all connections receive the guaranteed bandwidth.

In addition to call admission control, the CC also includes mechanisms for call termination control, which determine whether a call can be terminated given the current load on the network. These mechanisms ensure that the network resources are released fairly among all connections, preventing any single connection from monopolizing the network resources.

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The CC protocol is designed to ensure that all connections receive the guaranteed bandwidth, regardless of the type of traffic. This is achieved through the use of admission control mechanisms, which determine whether a new connection can be established given the current load on the network. These mechanisms ensure that the network remains balanced and that all connections receive the guaranteed bandwidth.
Support the dynamic call route.

The simulation scenario is also introduced.

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

4. The call blocking probability is in the range of 2.5 to 5.

eds (0).

descriptions of these calls can be drawn on a deterministic or stochastic.

The simulation environment.

Higher the amount of communication capacity required.

Data centers and switches and the network performance can be used to compute the capacity and efficiency of the network.

2. The lower the amount of capacity required by the network.

The lower the apparent bandwidth of a call, the higher is the amount of

3. The call duration model.

The simulation scenario is also introduced.

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Figure 1: Probability density of call duration generated by the simulator.

3.2 Voice Encoding

Simulator implementations ADPCM sources. ADPCM encoding for continuous rates of 4, 8, 16, and 32 Kbps. Output
rates of 8, 16, and 32 Kbps are produced by a specific ADPCM encoder. The encoded speech is used in various
encoding and decoding processes. The voice signal is sampled every
0.02 seconds, and each sample is encoded as a PCM value. The
encoded value, a 16-bit integer, is quantized to 1 of 256
levels. The encoded value is then converted to
a binary representation with a uniform probability distribution of
short calls (shorter than

\[ (x)f \cdot (m - 1) + (x)f \cdot m = (x)f \]

functions.
3.4 Cell Admission Control

The Cell Admission Control (CAC) function is responsible for determining whether a new cell can be added to the network without violating the network's resource constraints. The CAC process involves the following steps:

1. Admission Control Decision: The CAC function evaluates whether the incoming cell can be admitted into the network without exceeding the available resources. This decision is based on the current network state and the parameters of the incoming cell.

2. Admission Control Algorithm: The CAC function uses an admission control algorithm to make the decision. The algorithm considers factors such as the quality of service (QoS) requirements of the incoming cell and the current network load.

3. Admission Control Interface: The CAC function provides an interface for communicating with the network elements and the network control system. This interface allows the CAC function to interact with other network components and to receive feedback from the network.

4. Admission Control Monitoring: The CAC function continuously monitors the network state to ensure that the admission control decisions are accurate and up-to-date. This monitoring allows the CAC function to adapt to changes in the network environment and to respond to network failures.

3.5 Link model and Protocol Stack

The link model and protocol stack are responsible for the transport and delivery of data between network elements. The link model defines the interface between the network and the underlying physical medium, while the protocol stack defines the protocols used to exchange data between layers of the network.

The link model supports the transport of data by providing a reliable and efficient means of communication between network elements. The protocol stack provides the necessary services for data transport and delivery, including error control, flow control, and congestion control.

The link model and protocol stack are essential for the proper operation of the network. They ensure that data is transmitted efficiently and that errors are detected and corrected. The link model and protocol stack are also critical for ensuring the security of the network and for providing QoS guarantees to network users.

3.6 Logical Link Control (LLC) and Link Layer Protocol

The Logical Link Control (LLC) protocol is responsible for the logical transport of data between network nodes. The LLC protocol provides a common interface for upper-layer protocols and allows for the exchange of data between network nodes without regard for the underlying physical medium.

The LLC protocol supports the transport of data by providing a logical connection between network nodes. The LLC protocol uses a set of service access points (SAPs) to define the interface between upper-layer protocols and the LLC protocol itself.

The LLC protocol also provides error detection and correction services to ensure the reliability of the transmitted data. The LLC protocol uses a sliding window protocol to ensure that the data is transmitted efficiently and that the transmission errors are corrected.

3.7 Protocol Stack Architecture

The protocol stack architecture is responsible for the transport and delivery of data between network elements. The protocol stack is composed of multiple layers, each of which is responsible for a specific aspect of data transport and delivery.

The protocol stack architecture supports the transport of data by providing a common interface for upper-layer protocols and allowing for the exchange of data between network nodes without regard for the underlying physical medium.

The protocol stack architecture also provides error detection and correction services to ensure the reliability of the transmitted data. The protocol stack architecture uses a sliding window protocol to ensure that the data is transmitted efficiently and that the transmission errors are corrected.

3.8 Network Layer Protocol

The network layer protocol is responsible for the transport and delivery of data between network nodes. The network layer protocol provides a common interface for upper-layer protocols and allows for the exchange of data between network nodes without regard for the underlying physical medium.

The network layer protocol supports the transport of data by providing a logical connection between network nodes. The network layer protocol uses a set of service access points (SAPs) to define the interface between upper-layer protocols and the network layer protocol itself.

The network layer protocol also provides error detection and correction services to ensure the reliability of the transmitted data. The network layer protocol uses a sliding window protocol to ensure that the data is transmitted efficiently and that the transmission errors are corrected.
network model

The network model is based on a network of nodes, each of which is connected to other nodes through links. The performance of the network is determined by the performance of these links, which are represented by the connections between the nodes. The network model includes a wide range of features, such as the number and type of connections, the capacity of the links, and the delays associated with the transmission of data. The network model is used to simulate the behavior of the network under different conditions, allowing researchers to evaluate the performance of the network and make improvements as necessary.
Figure 4: Network topology used in the simulation.

Simulation Results

Figure 2: Example from the topology of a Circuit Switched Telephone Network.

Figure 3: Example from the topology of a Circuit Switched Telephone Network.
4.2. Packetization

The maximum utilization achievable in this scenario

The overhead of the VC is


![Graph showing packetization]

Voice over IP: Overallocation Overhead (Pack Delay 32ms)
Pack Delay 32 ms
(ADPCM 32)
Circuit Switching
Pack Delay 18 ms
(Voice over IP: Link Performance)

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

Voice over IP: Link Performance
the transport efficiency is considerably low.

The diagram on the right shows the blocking probability of the call over time. The curve represents the percentage of calls that are blocked due to exceeded capacity. The initial blocking probability is high, but it decreases over time as the network capacity is increased. The diagram on the left shows the network efficiency over time, indicating the percentage of efficient calls made. The network efficiency improves as the network capacity increases.

The network efficiency can also be monitored by studying the call blocking probability over time. A lower blocking probability indicates a more efficient network.

Figure 6: Blocking Probability and Real Load

Figure 7: Blocking Probability and Effective Load
The header size depends on the protocol and network configuration. Compared to the header overheads for different protocols, the header size of the packet, the delay, and the bandwidth of the network are the key factors affecting the overhead and delay.

**Figure 10** Impact of packetization delay over the real bandwidth of a phone call with different protocols.
4.3 Synchronous Optical Network (SONET) and Synchronous Digital Hierarchy (SDH) Compatibility

The primary objective of SONET/SDH is to provide a single channel from a central office to a remote office. This allows multiple channels to be transmitted simultaneously within a single fiber optic line. The SONET/SDH hierarchy is divided into three levels:

- OC-1 (51.84 Mbps)
- OC-3 (155.52 Mbps)
- OC-12 (622.08 Mbps)

The SONET/SDH protocol includes a number of features that ensure the reliability and integrity of the data transmitted. These features include

- Frame Synchronization
- Line Coding
- Path Monitoring

Packetization Overhead: the Apparent Bandwidth

The apparent bandwidth is the effective bandwidth available for data transmission. It is calculated as follows:

\[ \text{Apparent Bandwidth} = \text{Actual Bandwidth} - \text{Packet Overhead} \]

This formula helps in determining the effective bandwidth available for data transmission over the network.
5 Discussion

Figure 12: Impact of Packetization Delay on the Link Efficiency

Packetization Delay (ms)

Optimal Packet Delay

Effective Load (R[U])

The optimal packet size depends on many parameters. However, a common equation for packet we obtain:

\[
\text{Packet size (m) = } \left(1 + \frac{H}{D_{\text{peak}}} \right) \cdot \frac{D_{\text{peak}}}{D_{\text{peak}} - D_{\text{prop}}} - D_{\text{prop}}<m
\]

This equation is a good approximation when the loss rates high enough.
References

Acknowledgments

The main conclusion we can draw from the simulation results are:

to improve the efficiency of the network.

The performance of the system, the achieved efficiency does not apply.

to the efficiency of the network, the achieved efficiency does not apply.

to the efficiency of the network, the achieved efficiency does not apply.

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