On the Efficiency of Packet Telephony

Original

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On the Efficiency of Packet Telephony

Abstract

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The efficiency of packet telephony
introduced by each node in the network is referred to as the propagation delay. The propagation delay is defined as the number of time units required for a signal to travel from one node to another node in the network. The propagation delay is a function of the distance between the nodes and the speed of light.

The propagation delay is an important factor in determining the performance of a network. A network with a high propagation delay will have poor performance, as the delay will increase the time it takes for data to travel from one node to another node.

This delay can be written as:

$$\phi_d = \frac{t_d}{t_p}$$

where $t_d$ is the delay time and $t_p$ is the propagation time.

In a network with multiple hops, the total delay is the sum of the delay at each hop. This can be calculated as:

$$\sum_{i=1}^{n} \phi_i$$

where $\phi_i$ is the propagation delay at hop $i$ and $n$ is the number of hops.

This delay is an important factor in determining the performance of a network, as it affects the overall latency of the network. A network with a low propagation delay will have better performance than a network with a high propagation delay.
of the amount of real-time traffic.

2.3 Evaluating the Difference of Guaranteed Services

When a traffic flow is assigned a certain level of service, the QoS parameters, such as bandwidth and delay, are reserved in advance. However, when the traffic flow is not yet fully established, the QoS parameters are not guaranteed. Therefore, when a traffic flow is detected to be potentially exceed the available resources, the QoS parameters are renegotiated to ensure the service level is maintained.

2.4 Call Admission Control

The call admission control is essential for ensuring that the network resources are not over-subscribed. When a new call is received, the network checks if there are sufficient resources to support the call. If the resources are available, the call is admitted; otherwise, the call is rejected.

In practice, the call admission control can be implemented in various ways, such as using a queuing mechanism or a resource reservation protocol. The decision on whether to accept or reject a call is based on a set of predefined rules or policies.
The network model and the number of calls offered to the network

The effective load drops in the range of mix, of calls.

The call blocking probability is the ratio between the number of calls

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3.2 Voice Encoding

Simulators are employed to determine the probability density function of the signal to be encoded. The signal is first processed by the simulator to generate a probability density function. This function is then used to encode the signal, resulting in a coded signal.

The encoded signal is then transmitted over the communication channel. During transmission, the signal may be subject to various forms of interference, such as noise and distortion. To mitigate these effects, error correction techniques are employed. These techniques involve the use of error-correcting codes, which are designed to detect and correct errors that may occur during transmission.

Error correction techniques are essential in ensuring the reliability of the transmitted signal. These techniques are particularly important in applications that require high-quality audio reproduction, such as voice communication systems.

In summary, the process of voice encoding and transmission involves the use of simulators to generate a probability density function, which is then encoded and transmitted over the communication channel. Error correction techniques are employed to ensure the reliability of the transmitted signal.
3.3 Link model and Protocol Stack

- **3.3.1 Link model**
  - A protocol or a layer (like the protocol hierarchy) can be either a layer or a protocol.
  - The layer protocol provides an interface for the application layer, which is used to transmit data between the layers.
  - The protocol model is used to define the interface between the layers.

- **3.3.2 Protocol Stack**
  - The protocol stack is a collection of protocols that work together to provide a unified interface for the application layer.
  - It consists of several layers, each responsible for a specific aspect of communication.
  - The layers are ordered from bottom to top, with each layer protocol providing a service to the layer above it.
  - The layers are: Physical, Data Link, Network, Transport, Session, Presentation, and Application.

- **3.3.3 Link Control**
  - The link control layer is responsible for managing the physical link between devices.
  - It handles the establishment, maintenance, and termination of the link.
  - It also performs error detection and recovery.

- **3.3.4 Protocol Control**
  - The protocol control layer is responsible for managing the communication between the layers.
  - It handles the exchange of data between the layers, including the transfer of packets.
  - It also performs error detection and recovery.

- **3.3.5 Link Layer Protocol**
  - The link layer protocol is responsible for managing the actual transmission of data over the physical link.
  - It handles the conversion of data into a format suitable for transmission and the reverse process.
  - It also performs error detection and recovery.

- **3.3.6 Protocol Layer Protocol**
  - The protocol layer protocol is responsible for managing the communication between layers using the protocol control layer.
  - It handles the exchange of data between the layers, including the transfer of packets.
  - It also performs error detection and recovery.

- **3.3.7 Link Layer Control**
  - The link layer control is responsible for managing the actual transmission of data over the physical link.
  - It handles the conversion of data into a format suitable for transmission and the reverse process.
  - It also performs error detection and recovery.

- **3.3.8 Protocol Layer Control**
  - The protocol layer control is responsible for managing the communication between layers using the protocol control layer.
  - It handles the exchange of data between the layers, including the transfer of packets.
  - It also performs error detection and recovery.

3.4 Call Admission Control

- **3.4.1 Call Admission Control**
  - A call admission control (CAC) protocol is used to determine whether a call can be accepted on a given link.
  - It is used to prevent the network from becoming congested.
  - The CAC protocol uses various algorithms to determine whether a call can be accepted.

- **3.4.2 CAC Algorithms**
  - There are several CAC algorithms, including the following:
    - **Token Bucket Algorithm**
    - **RED (Random Early Detection)**
    - **DQDB (Double-Queue Delayed Backpressure)**
    - **CBR (Constant Bit Rate)**
    - **PBB (Preliminary Bandwidth Balance)**

- **3.4.3 CAC Implementation**
  - The CAC algorithm is implemented in the network layer of the protocol stack.
  - It uses various metrics, such as link utilization and queue length, to determine whether a call can be accepted.
  - The CAC algorithm is designed to be scalable and adaptable to different network conditions.

3.5 Link Control and Protocol Stack

- **3.5.1 Link Control**
  - The link control layer is responsible for managing the actual transmission of data over the physical link.
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To ensure that the model results obtained have satisfactory performance, we consider a network model in which each node is connected to a local network model by exchanging information through the subnetworks. We consider a network model in which each node is connected to a local network model by exchanging information through the subnetworks.
network topology used in the simulation.

Simulation Results

The network topology used in the simulations (see Figure 4) has been modified as a result of feedback from users and local authorities. The modifications have been implemented to improve network performance and reliability. The feedback from users and local authorities has been taken into account to make necessary adjustments to the network topology.

The modified network topology is shown in Figure 4. The modifications include changes in the placement of network nodes and the introduction of new links to improve connectivity and reduce network latency. The network is designed to handle high traffic volumes and provide reliable service to users in the region. The modifications have been made to ensure that the network is scalable and can accommodate future growth.

The simulation results show a significant improvement in network performance compared to the original topology. The modifications have resulted in reduced network latency, increased capacity, and improved reliability. The network is now able to handle higher traffic volumes without degradation in service quality.

In conclusion, the modifications to the network topology have been successful in improving network performance and reliability. The network is now better equipped to handle high traffic volumes and provide reliable service to users in the region. The modifications have been made with the goal of ensuring that the network is scalable and can accommodate future growth.
4.2 Packet Over-allocation

compared as shown in the next section. However, it is important to note that the packets are only over-allocated if the difference between the apparent load and the effective load is negative.

The difference between the apparent load and the effective load is determined by the over-allocation factor, which is the ratio of the apparent load to the effective load. The over-allocation factor is calculated as the difference between the apparent load and the effective load divided by the effective load. The over-allocation factor is then used to adjust the apparent load to account for the over-allocation.

4.1 Bandwidth Over-allocation

Dotted line: Efficient allocation of link T0 - T1, with high packet allocation.

Overall result is that the packet over-allocation is a good representation of the real load. This is shown by the apparent load (T0 - T1) and the real load (T0). In this scenario, the packet over-allocation factor (POF) is used to determine the over-allocation factor. The over-allocation factor is then used to adjust the apparent load to account for the over-allocation.
4.2. The Payload

a) Utilization Efficiency:

The utilization efficiency of the link is one of the two parameters on the real-time network between the effective and real loads. This section presents a quantitative description of the impact of packet size over the efficiency of effective load.

Packet Delay: Impact of packet size over the efficiency of effective load.

Offered Load (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)

0% 20% 40% 60% 80%

Effective Load

Effort of Packets over the Link: Performance

Offered Load (Erlang)

Voice over IP: Link Performance

Voice over IP: Overallocation Overhead (Pack Delay 18ms)
The transport efficiency is considerably low. Once the ADPCM32 frame is received, the transport efficiency is not recognized as the best solution. In contrast, a shorter retransmission delay is not necessary for the best solution. In other words, it is believed that retransmission delay is not necessary for the transmission of best effort traffic. Thus, the retransmission delay of best effort traffic is identical to that of real-time traffic.

Blocking probability is shown for the second and third cases. The second case corresponds to a retransmission delay of 18 ms, and the third case to a circuit switched network. Since the blocking probability for the second case is lower than that of the third case, the efficiency of the second case is higher than that of the third case. In this case, the network performance is affected by the call blocking and the network efficiency.

Figure 6 shows the blocking probability and real-load distribution for the second case. The blocking probability is calculated by the call blocking and the network efficiency.

The retransmission delay is smaller in the second case. In this case, the network performance is also affected by the call blocking and the network efficiency. In the second case, the call blocking is lower than that of the third case. Thus, the network efficiency is higher in the second case.
In a switched network, the real path is made up of the actual transmission path the call takes from the source to the destination, and the performance of the network depends on the quality of the transmission path. The performance of a switched network can be affected by factors such as congestion, latency, and packet loss. Therefore, it is important to design and manage a switched network to ensure that it meets the required performance levels.

**Figure 1:** Impact of packetization delay over the real bandwidth of a phone call with various technologies.

When packetization delay is encountered, the optimal number of samples per packet is determined by the maximum number of samples that can be transmitted in a single packet. In a circuit-switched network, each packet carries a fixed number of samples, and the delay is determined by the time it takes to transmit the packet. However, in a packet-switched network, the delay is determined by the time it takes to store the packet in memory and then transmit it.

**Circuit Switching (ADPCM32):**

- **Voice over IP over SONET:**

**Circuit Switching (PCM64):**

- **Voice over ATM:**

When the packetization delay is longer than the delay threshold, the quality of the call is affected. Therefore, it is important to design and manage a switched network to ensure that it meets the required performance levels.
4.3 SONEF/SDH and Voice Compression

The primary objective of the design is to achieve a cost-effective implementation of the voice compression technology. This is achieved by minimizing the use of voice compression codecs, while ensuring a high-quality voice service. The design utilizes a combination of SONEF/SDH and voice compression technologies to achieve this goal.

The SONEF/SDH technology is used to transmit voice data over the SONEF/SDH network. The voice data is then compressed using a voice compression codec. The compressed voice data is then transmitted over the SONEF/SDH network. At the receiving end, the voice data is decompressed and sent to the appropriate destination.

The diagram below illustrates the impact of the voice compression technology on the apparent bandwidth of a voice call over a SONEF/SDH network.
5 Discussion

The measured performance of the proposed delay function can be characterized by the equation

\[ D_{\text{prop}} = \frac{I + H}{D_{\text{req}} - D_{\text{prop}}} \]

where

- \( I \) is the number of hops required by a packet to travel from one node to another in the network,
- \( H \) is the header size of the packet, and
- \( D_{\text{req}} \) is the required delay for the packet.

The optimal packet size depends on many parameters. However, a common

equation for the case of delay can be obtained

\[ D_{\text{prop}} = \frac{I + H}{D_{\text{req}} - D_{\text{prop}}} \]

To account for some of the delay components in (6),

- the optimal packet size is defined as the packet size that minimizes the delay parameter.
The main conclusion we can draw from the simulation results are:

- The reference model can work as expected under different conditions.

In order to simplify the implementation of the reference models, the state transitions caused by packet arrivals, there is no reason to invest in the provision of state data in the network. However, if the user is satisfied with the available real-time requirements, the efficiency of state transitions can be improved by storing state information periodically. The efficiency of state transitions can be improved by storing state information in a memory that is available for easy access. The state transitions can be improved by storing state information in a memory that is available for easy access.