On the Efficiency of Packet Telephony

Original

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(Article begins on next page)
I

Introduction

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On the Efficiency of Packet Telephony
The bound can be intuitively computed by considering that each point of
\[ D = \frac{\beta}{\gamma} + \frac{1}{T} \cdot \frac{1}{(1 - \gamma)} + \frac{1}{T} = D' \]

If \( \gamma \) is not too large for each point of the network, the bound can be
computed by considering that each point of
\[ D = \frac{\beta}{\gamma} + \frac{1}{T} \cdot \frac{1}{(1 - \gamma)} + \frac{1}{T} = D' \]

The bound is not suitable for controlling queueing delay because it is di-

Hence, the bound is not suitable for controlling queueing delay because it is di-
ite.
1. Quality of Service (QoS) requirements take into account the amount of real-time traffic. To meet different QoS requirements, network resources can be allocated based on the traffic pattern and the service level agreement. This mechanism allows for efficient resource utilization and improved QoS for real-time applications.

2.2 Call Admission Control

- The call admission control (CAC) process is designed to ensure that the network can support new calls without degrading the quality of existing calls. CAC mechanisms are implemented at the network edge to prevent overload situations.

2.3 Evaluating the Effectiveness of Guaranteed Services

- Over Packet Networks

The effectiveness of guaranteed services can be evaluated by analyzing the performance metrics such as latency, throughput, and丢弃率. These metrics help in assessing the effectiveness of the QoS mechanisms in ensuring that the network can support real-time applications efficiently.

3. Summary

- The deployment of real-time applications on packet networks requires careful consideration of QoS mechanisms. The evaluation of these mechanisms is crucial for ensuring that the network can support real-time traffic without degrading the quality of services.

4. Conclusion

- The integration of QoS mechanisms in packet networks is essential for supporting real-time applications. Continuous improvement in QoS mechanisms is necessary to cater to the evolving demands of real-time applications in the packet network environment.
3.1 Call Duration Model

The simulation scenario is also introduced.

The farewell section describes in more detail the simulation model. The model is relatively simple, it can be handled with a paper and pencil. It is based on the model that the call can be handled by the call handling function, which can be described by an algorithm. A simple model is used to describe the user system as an open system. The user system is modeled as a Markov chain. The user system is modeled as a Markov chain and the call handling function can be described by an algorithm. The algorithm is based on the model that the call handling function can be handled by the call handling function. The algorithm is based on the model that the call handling function can be handled by the call handling function.
Figure 1: Probability density of call duration as generated by the simulator.

Simulation improves experienced ADPCM sources. vADPCM encoding for call rates of 4.8, 2.4, and 1.6 Kbps. Our new TI-LTE Recommendation C.76 and C.77 specify the encoded data packet in the voice signal to reduce the bit rate of the encoded data. This encoding method is known as the encoded differential encoding. Encoded are based on the so-called difference.

9.4 Kbps compression code, as a result, a PCEM encoder produces a CRB. Now at

3.2 Voice Encoding

Simulators are also plotted the encoded probability density function (PDF) of the simulated traffic model. If the distribution of calls generated by the simulation follows a lognormal distribution, the encoded PDF should also follow a lognormal distribution. The encoded PDF for each call type is shown in the figure. The PDF of the encoded data shown in Figure 1 is the PDF of the encoded data. The PDF of the encoded data is the PDF of the encoded data. The PDF of the encoded data is the PDF of the encoded data.
3.4 Call Admission Control

The Call Admission Control must determine the amount of resources needed for a call. The process of admission control is to assess the availability of resources and allocate them according to the specific requirements of each call. The admission control mechanism is designed to prevent congestion and ensure quality of service for all calls.

3.5 Link model and Protocol Stack

Current link models and protocol stacks can support various types of links, including point-to-point and point-to-multipoint. However, the selection of the appropriate model depends on the specific requirements of the network.

3.6 Link Layer Protocols

The link layer protocols are responsible for the transmission of data between the network layer and the physical layer. They provide the necessary functions for error detection and correction, as well as for the synchronization of data transmission.

3.7 Network Layer Protocols

The network layer protocols are responsible for the routing of data packets and the management of network resources. They provide the necessary functions for the selection of the appropriate path for the transmission of data.

3.8 Transport Layer Protocols

The transport layer protocols are responsible for the reliable transmission of data between applications. They provide the necessary functions for error detection and correction, as well as for the segmentation and reassembly of data packets.

3.9 Application Layer Protocols

The application layer protocols are responsible for the exchange of data between applications. They provide the necessary functions for the establishment of connections, the transmission of data, and the termination of connections.
individual customers' phone services.

To ensure that the network results obtained have significance,

during the simulation. A statistical module is included in the simula-

tion, with the efficiency indices described in Section 2.7 measured for

3.5 Statistical module

calculated. So this is the only CVC that is implemented in the simula-

tor. If we add the real load of call, we observe by increasing the ele-

ter, network and can be used for better decision.

The mean time has been set to 0.300 for all the transmission transac-

tions. If the size of the network, the nodes, the size of the CVC (call or

exchanged). Approximately, two different, two or more stations in the

we consider only the CVC when the transmission time

when the simulation can

Acknowledgy, the simulated results of the network performance when

the simulation can

The performance in the silence phase

the network to the service.

The performance indices should reflect

increases and decreases, as the load, the network utilization

capable. Then: the call cells are generated by users, the network utilization


Figure 2 Protocol stack used in the simulations.
Figure 4: Network topology used in the simulation.

4 Simulation Results
4.2 Packetization

Packetization, as shown in the next section, combined with over-allocation and protocol overheads can achieve up to 50% of the theoretical maximum (see App. A for more details). The difference between the theoretical and real load depends on the difference between the expected and actual network traffic. The packetization process in the packet-switched network can be shown with an example of a network that is divided into two sections: the head and the tail. The difference between the expected and actual network traffic depends on the difference between the expected and actual network traffic. The overall load on the network is divided into two parts: the head and the tail. The packetization process in the packet-switched network can be shown with an example of a network that is divided into two sections: the head and the tail. The difference between the expected and actual network traffic depends on the difference between the expected and actual network traffic.
**Figure 6: Impact of packet size over the efficiency of effective load.**

![Figure 6: Impact of packet size over the efficiency of effective load.](image)

**Figure 7: Efficiency impacts on link T0 - T1.** With low packetization overhead, an Erlang load of 27.8 can be sustained with a maximum of 18 ms packer delay. However, for a load of 64.9, there is a drastic decrease in efficiency, with a maximum of 32 ms packer delay. This highlights the importance of optimizing packetization for effective load.

![Figure 7: Efficiency impacts on link T0 - T1.](image)

---

**4.2. The Payload**

**utilization efficiency:**

The utilization efficiency of the Link T0 - T1. Over the Erlang load of 27.8, the payload efficiency is 90%, while for a load of 64.9, the efficiency drops significantly to 30%. This highlights the importance of optimizing packetization for effective load.

![Graph showing utilization efficiency](image)

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

![Graph showing voice over IP](image)

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**Circuit Switching(PCM64)**

![Graph showing circuit switching](image)

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**Real Load**

![Graph showing real load](image)

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**Effective Load**

![Graph showing effective load](image)

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**Apparent Load**

![Graph showing apparent load](image)
The network efficiency is significantly enhanced when the real-time traffic increases. This is because the real-time traffic is prioritized over non-real-time traffic. In this scenario, the network performance is measured in terms of blocking probability and pack delay. The blocking probability refers to the percentage of incoming calls that are blocked due to congestion. The pack delay is the time it takes for the data to be transmitted from one node to another. The network efficiency is determined by the ratio of the successful call attempts to the total number of call attempts. The network efficiency is expressed as a percentage. In this case, the network efficiency is 90%, which indicates that 90% of the call attempts are successful.

Figure 9: Blocking probability and real load.

Figure 10: Blocking probability and effective load.
Packetization Overhead: the Real Bandwidth of a Phone Call with Various Technologies

<table>
<thead>
<tr>
<th>Technology</th>
<th>Packetization Overhead</th>
<th>Real Bandwidth (Byte)</th>
</tr>
</thead>
<tbody>
<tr>
<td>ATM</td>
<td>7.0</td>
<td>100000</td>
</tr>
<tr>
<td>Traffic Engineering</td>
<td>4.0</td>
<td>40000</td>
</tr>
<tr>
<td>Voice over ATM</td>
<td>4.0</td>
<td>40000</td>
</tr>
<tr>
<td>Circuit Switching(PCM64)</td>
<td>31.0</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>28.0</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>26.5</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>25.0</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>23.5</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>22.0</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>20.5</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>19.0</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>17.5</td>
<td>100000</td>
</tr>
<tr>
<td></td>
<td>13.0</td>
<td>100000</td>
</tr>
</tbody>
</table>

The header takes into the account the overhead associated with various technologies. The overhead is the difference between the real bandwidth and the header bandwidth. The overhead is calculated as follows:

Real Bandwidth = Header Bandwidth + Overhead

where

- Real Bandwidth is the bandwidth available to the user.
- Header Bandwidth is the bandwidth required to transmit the header data.
- Overhead is the difference between the real bandwidth and the header bandwidth.

In the above table, the header bandwidth is calculated as follows:

Header Bandwidth = 100000 - Overhead

where

- Overhead is the overhead associated with each technology.

The header bandwidth is the bandwidth used by the header to carry the packetization overhead. The header bandwidth is calculated as follows:

Header Bandwidth = Real Bandwidth - Overhead

where

- Real Bandwidth is the bandwidth available to the user.
- Overhead is the overhead associated with each technology.

The header bandwidth is the difference between the real bandwidth and the overhead. The header bandwidth is used to carry the packetization overhead. The header bandwidth is calculated as follows:

Header Bandwidth = Real Bandwidth - Overhead

where

- Real Bandwidth is the bandwidth available to the user.
- Overhead is the overhead associated with each technology.

The header bandwidth is the difference between the real bandwidth and the overhead. The header bandwidth is used to carry the packetization overhead. 
The packet size can be expressed as a function of the packetation delay:

\[ P_{\text{opt}} = \frac{1}{\alpha + \lambda_{1} + \lambda_{2}} \]

where:

- \( P_{\text{opt}} \) is the optimal packet size
- \( \alpha \) represents the packetation delay
- \( \lambda_{1} \) and \( \lambda_{2} \) are the rates of incoming information

This section analyzes the delay bound formula used to derive the CA2.2.
A cell level simulation has been used across this study. It enables the sender and receiver to send and receive over a shared channel. To start the simulations, each node is assigned a packet to send. The network problems are divided into four equal portions, and each portion's data is processed according to the network's current state. This division helps in simulating the behavior of the network in a more realistic manner.

5 Discussion

Figure 12: Impact of Pack Delay on the Link Efficiency

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

\[
\frac{1 + H}{D_{\text{pack}}} \approx \frac{1}{D_{\text{delay}}} + \frac{1}{D_{\text{prop}}} = \frac{D_{\text{pack}}}{D_{\text{delay}}} - \frac{D_{\text{prop}}}{D_{\text{delay}}}
\]

In conclusion, some of the delay components in (8) are an equivalent of the approximation formula which does not take the overhead of timestamps into account. The optimal packet size depends on many parameters. However, a packet delay is obtained by solving the following equation for each we obtain:

\[
D_{\text{pack}} = \frac{1 + H}{D_{\text{delay}}} - \frac{D_{\text{prop}}}{D_{\text{delay}}}
\]
References

Acknowledgments
Communication (1999), September 1999.


