

# IMPLEMENTATION OF AN OPTIMIZED PACKET SWITCHING PARAMETERS IN WIRELESS COMMUNICATION NETWORKS

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## Abstract

Network attributes that have profound effect on the perceived network performance are mainly targets of network modelling. These attributes represent the goals of the statistical analysis, design and optimisation of the network. Fundamentally, network models are constructed by defining the statistical distribution of the arrival and service rates in a queuing system that subsequently determine these attributes. But, the vague and imprecise nature of most of these attributes renders statistical simulation obsolete and calls for an intelligent approach. Nevertheless, the network is not the sole driver of data transfer speed and of the end-user experience. Many other factors directly impact how fast application queries and responses will flow through the network. If one wants to troubleshoot performance degradations, this checklist of the factors that can badly impact the transfer speed will come in very handy. The study is to develop an intelligent system for packet switching in wireless communication networks. The proposed system guarantees improved quality of service. With the implementation of the underlying concept of linguistic variable definitions for variables with uncertainties has been widely unrecognized—a fact which relates to the concept of precision. This Thesis offers precise/accurate solution to the packet switching problem, as opposed to non-linguistic specifications which produces unreliable solutions.

**Keywords:** Network Attribute, Throughputs, Latency, Packet Switching etc.

## INTRODUCTION

Packet Switched Networking (PSN) falls under Delay Tolerant Networking (DTN). The delay Tolerant Networking addresses the architectural and protocol design principles arising from the need to provide interoperable communications with and among extreme and performance-challenged environments where continuous end-to-end connectivity cannot be assumed.

Every IT Operations team faces complaints indicating that the network is slowing down or delivering applications poorly. Their very first focus is on verifying network performance factors (i.e., latency, packet loss, Packet arrival rates, traffic Intensity and Transmitted Packet length etc.).

Nevertheless, the network is not the sole driver of data transfer speed and of the end-user experience.

Many other factors directly impact how fast application queries and responses will flow through the network. If one wants to troubleshoot performance degradations, this checklist of the factors that can badly impact the transfer speed will come in very handy.

### (i) Network latency

Network latency refers to the time needed to send a packet from the source to the destination. This time varies depending:

- on the physical distance
- the number of network devices which have to be crossed (also referred to as the number of hops)
- and to a lesser extent, to the performance of each of the devices.

The relationship between latency and transfer rates depends on the protocol that carries the data. Maintaining our focus on the most common ones: for a UDP (user datagram protocol) flow, latency may not have an impact. As for TCP applications, typically the most commonly used protocol, it will have a drastic impact. And because of this enabling technologies are therefore required in order to support the challenges of PSN.

### (ii) Network congestion

Network congestion refers to the saturation of a path used by packets to flow between the source and the destination. The element on the path can be either an active device (e.g., router or switch) or a physical link (e.g., cable).

When the maximum capacity of the element is reached, the packet cannot be transferred in a timely manner as it is either put in a queue (e.g., in a router) or dropped if a no queue system is available to retain them. It may even become impossible to set up new sessions.

The consequence will then vary, depending on the level of delay generated by the congestion: Packets are delayed for a short period of time. The latency will increase, some retransmissions will occur (for TCP flows) as the acknowledgment packets are not received fast enough by the sender. Duplicate acknowledgment packets will also be received, Packets are lost or dropped (packet loss).

The retransmission increases significantly: as packets are not acknowledged, they will then be massively re-sent. Disconnections: sessions are dropped as too many packets are lost.

Random models are inappropriate as there is no structure for intelligent forwarding algorithms to exploit. Real systems must be built, measured, and learned from in

order to make progress on this most important facet of PSN.

**(iii) Infrastructure parameters (QoS, Filtering, Routing)**

Although the overall network path is free of any congestion (lack of bandwidth or system resources), some devices apply policies:

- **Prioritization:** some traffic is either more strategic (critical applications) or more performance sensitive (real-time applications, VoIP, video conferencing) and gets allocated a higher priority than the rest of the applications using a given network path. In case the maximum capacity on the network path is reached, lower priority flows will start experiencing retransmission, packet loss or disconnection depending on how long and important the congestion is.
- **Filtering/encryption:** there may be many kinds of filtering in place to scan viruses, to prevent users to reach non-recommended sites, to prevent threats to web servers, etc. Filtering has an impact on data transfer, depending on how much processing time it requires. This might have an impact on the latency between the client and the server. If the processing time becomes excessive, it can generate retransmission and packet loss.
- **Routing/load balancing:** some devices distribute the load across a group of servers/devices or route the traffic to an adequate path from a performance and/or an economic standpoint. The devices may also be overloaded or misconfigured which could lead to retransmission, packet loss or disconnection issues.
- **While troubleshooting slow transfer rates,** it is important to list the devices on the path between clients and servers. You can then identify at which point in time and for which flow: retransmissions, duplicate acknowledgments, packet loss, TTL (Time to leave) expired and session time-out or incomplete TCP start can be observed.

As a result of the above mentioned challenges an intelligent system must be developed that will enhance effective Packet switching in the wireless communication Networks

**(iv) Client or Server health**

It is probably the last item you will consider if you are focused on network performance. But these systems also have limited resources which can lead to a congestion situation and slow down the data transfer rates. If a server lacks hardware resources, such as RAM, CPU, I/O, it will process user queries slower. At a given moment, a client or a server reaching a congestion point will slow down the transfer using standard TCP mechanisms. Enabling technologies are therefore required in order to support the challenges of PSN.

**Literature Review**

Networks are quickly shifting from voice-centric to data-centric. In this developing scenario, the role of synchronous optical network/synchronous digital hierarchy (SONET/SDH) is likely to decrease, and the optical transport network will provide a transport infrastructure for legacy and new IP services directly. In addition, the use of wavelength-division multiplexing (WDM) offers massive bandwidth through the parallel transmission of high-bit-rate channels on the same fibre, at very attractive costs per bit. Currently, researchers are witnessing the shift from purely point-to-point WDM

systems to switching systems with real networking functionality at the optical level. Additionally, because of the migration of most services over IP and the evolution of optical circuit switching, core networks tend to have a two-level IP-over-optical architecture. Optical cross-connects (OXC), however, offer granularity only at the wavelength level. For networks that must provide a large variety of service levels in a highly dynamic environment, this has an impact on the dimensioning of the network and the size of the OXCs.

Packet switching (PS) offers the flexibility and bandwidth efficiency called for in this environment. Current (PS) networks transport IP packets in the optical domain but convert the packets to the electronic domain to route them at intermediate nodes.

When a packet arrives at a router in a datagram network, the only information in the packet that can help the router in its routing is the destination address of the packet. The table then is sorted to make the searching faster. Today's routers use some sophisticated searching techniques. When a packet arrives at a switch in a virtual circuit network, the pair (input port, input VCI= Virtual Circuit Identifier) can uniquely determine how the packet is to be routed; the pair is the only two pieces of information in the packet that is used for routing. The table in the virtual-circuit switch is sorted based on this pair. However, since the number of port numbers is normally much smaller than the number of virtual circuits assigned to each port, sorting is done in two steps: first according to the input port number and second according to the input VCI (Murthy & Groomer, 2001).

A datagram or virtual-circuit network handles packetized data. For each packet, the switch needs to consult its table to find the output port in the case of a datagram network, and to find the combination of the output port and the virtual circuit identifier in the case of a virtual-circuit network. In a circuit-switched network, data are not packetized; no routing information is carried with the data. The whole path is established during the setup phase.

(iii)According to Clos,  $n = (N/2)^{1/2} = 7.07$ . the researcher can choose  $n = 8$ . The number of crossbars in the first stage can be 13 (to have similar crossbars). Some of the input lines can be left unused. The researcher then have  $k = 2n - 1 = 15$ . Figure 1 shows the configuration.

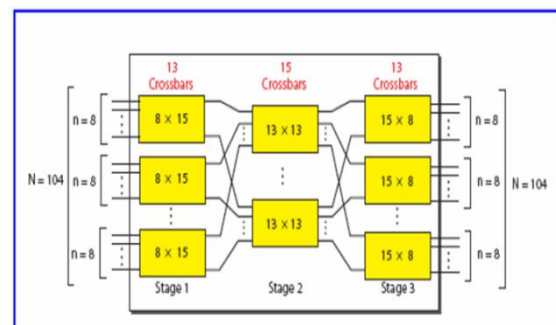


Fig.1: Configuration of Crossbars (Source: Larry Roberts. The Evolution of Packet Switching, invited paper, IEEE, November 1978)

The research can calculate the total number of cross-points as:

$$13 (8 \times 15) + 15 (13 \times 13) + 13 (15 \times 8) = 5655$$

The number of cross-points is still much less than the case with one crossbar (10,000). The researcher sees that there is no blocking involved because each 8 input line has 15 intermediate cross-bars. The total number of cross-points here is a little greater than the minimum number of cross-points according to Clos using the formula  $4N [(2N)^{1/2} - 1]$ , which is 5257.

- (a) If  $n > k$ , an  $n \times k$  crossbar is like a multiplexer that combines  $n$  inputs into  $k$  outputs. However, we need to know that a regular multiplexer is  $n \times 1$ .
- (b) If  $n < k$ , an  $n \times k$  crossbar is like a demultiplexer that divides  $n$  inputs into  $k$  outputs. However, we need to know that a regular demultiplexer is  $1 \times n$ .
- (iv) Routing consider the following conditions:
  - (a) In a circuit-switched network, end-to-end addressing is needed during the setup and teardown phase to create a connection for the whole data transfer phase. After the connection is made, the data flow travels through the already-reserved resources. The switches remain connected for the entire duration of the data transfer; there is no need for further addressing.
  - (b) In a datagram network, each packet is independent. The routing of a packet is done for each individual packet. Each packet, therefore, needs to carry an end-to-end address. There are no setup and teardown phases in a datagram network (connectionless transmission). The entries in the routing table are somehow permanent and made by other processes such as routing protocols.

First:  $(3200 \text{ Km}) / (2 \times 108 \text{ m/s}) + (3 + 20 + 20) = 59.0 \text{ ms}$

Second:  $(11700 \text{ Km}) / (2 \times 108 \text{ m/s}) + (3 + 10 + 20) = 91.5 \text{ ms}$

Third:  $(12200 \text{ Km}) / (2 \times 108 \text{ m/s}) + (3 + 10 + 20 + 20) = 114.0 \text{ ms}$

Fourth:  $(10200 \text{ Km}) / (2 \times 108 \text{ m/s}) + (3 + 7 + 20) = 81.0 \text{ ms}$

Fifth:  $(10700 \text{ Km}) / (2 \times 108 \text{ m/s}) + (3 + 7 + 20 + 20) = 103.5 \text{ ms}$

- (c) In a virtual-circuit network, there is a need for end-to-end addressing during the setup and teardown phases to make the corresponding entry in the switching table. The entry is made for each request for connection. During the data transfer phase, each packet needs to carry

a virtual-circuit identifier to show which virtual-circuit that particular packet follows.

- (v) A packet switch has four components: input ports, output ports, the routing processor, and the switching fabric. An input port performs the physical and data link functions of the packet switch. The output port performs the same functions as the input port, but in the reverse order. The routing processor performs the function of table lookup in the network layer. The switching fabric is responsible for moving the packet from the input queue to the output queue.
- (vi) There are two approaches to packet switching: datagram approach and virtual circuit approach
- (vii) The address field defines the virtual circuit number (local) addressing.
- (viii) The address field defines the end-to-end (source to destination) addressing
- (ix) Switching provides a practical solution to the problem of connecting multiple devices in a network. It is more practical than using a bus topology; it is more efficient than using a star topology and a central hub. Switches are devices capable of creating temporary connections between two or more devices linked to the switch. (Source: Accounting Information Systems: A Database Approach by: Uday S. Murthy, Ph.D., ACA and S. Michael Groomer, Ph.D., CPA, CISA. July 20, 2001. Copyright © 1996-2001 Cyber Text Publishing, Inc. All Rights Reserved)

### 5.3. INTERACTION BETWEEN PACKETS SWITCHING PARAMETERS

In this section we investigate the interaction between the packet switching parameters under study. The aim is to study the relationships that exist between the various parameters and how one parameter can be exploited for the benefit of the other parameters, to ensure good system performance.

#### 5.3.1 INTERACTION BETWEEN LATENCY AND PACKET ARRIVAL RATE

The packet delay (latency) of the arrival packets, sequentially aggregated over time is demonstrated in Figure 30. It is clear that there is long packet aggregation delay when the packet arrival rate is very low. This implies that the first arrival packet should wait longer time in the buffer before the number of packets in the buffer reaches the packet aggregation threshold. It is also obvious that with the same arrival rate, there is higher aggregation delay with bigger threshold of packet aggregation. On the other hand, the channel access delay increases with increasing packet arrival rate. Therefore, the packet delay decreases first when packet arrival rate increases due to the effect of decreasing aggregation delay. The packet delay will slightly increase because of the effect of the increasing channel access delay when the packet arrival rate getting higher. Further,

when the packet arrival rate is low, it cannot achieve throughput gain by packet aggregation, due to high delay expense. Therefore it is not worth doing packet aggregation for instance when packet arrival rate is lower than 10 in this case. But when packet arrival rate is high, it can achieve phigher throughput through packet aggregation with low packet delay. This packet aggregation delay can also be extended to send packets before the first packet becomes moribund. So it is a trade-off between the throughput and the delay. This can guarantee the minimum throughput with fixed number of packets to aggregate. But with respect to practical applications and services requirements, the number of aggregated packets should be adaptive according to the delay constraint.

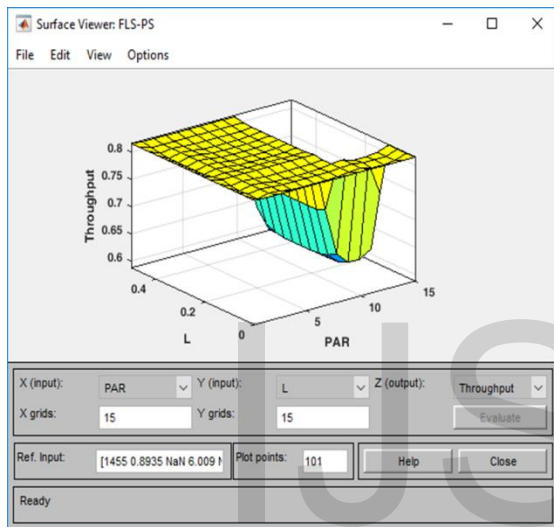


Figure. 30: Interaction between latency and packet arrival rate Low L vs. High PAR

### 5.3.2. Interaction between Latency and Packet Loss

Figure. 31. introduces the packet loss, on the left side we have 0% loss and no latency so we get almost full bandwidth utilisation. When packet losses are introduced, the system experiences congestion and struggles to fill the link again even without latency impact. If we then introduce 0.2 of round-trip latency the throughput is impacted by congestion control, slow start and leads to low link utilization. Hence, further simulation with packet loss and higher latency above 0.2 completely degrades the throughput.

In practical systems, exponential is the worst case, and the linear is the best. In general when one or more packets are lost or damaged, they are resent, leading to a drop in throughput equal to the size of the packet payload/resent time. But if the resend itself fails and the packet must be resent again, the trend can approach exponential.

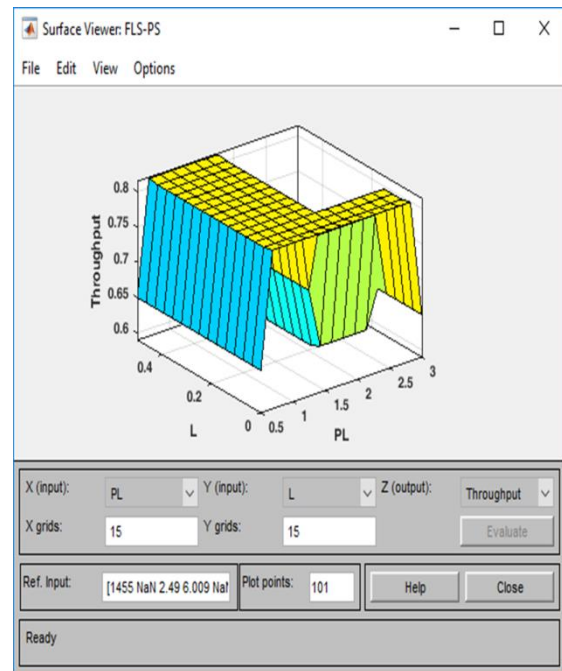


Figure.31: Interaction between latency and packet loss

### 5.3.3 Interaction between Latency and Traffic Intensity

With heavy-loaded traffic, the utilization levels of the channel access Opportunities (transmission resources) provided by the underlying network are high (i.e., the traffic intensities are high). We observe from Fig.32. That the proposed system achieves about 80% of the maximum system throughput (which is achieved with heavy-loaded traffic), while reducing the delay to half or less compared to the heavy-loaded traffic. The throughput reduction may be due to some missed transmission opportunities, i.e., transmission opportunities that are assigned to a node but cannot be utilized since there is no relay traffic and no source input traffic available for transmission. Hence, with the input traffic control, the utilization level of the network transmission resources is lower than with the heavy-loaded traffic. This delay reduction can be attributed to the somewhat lower traffic loads that result from the control of the input traffic. Put differently, the lower utilization levels of the network transmission resources, i.e., the lower traffic intensities, with the input traffic control result in lower mean packet delays than with the heavy-loaded traffic.



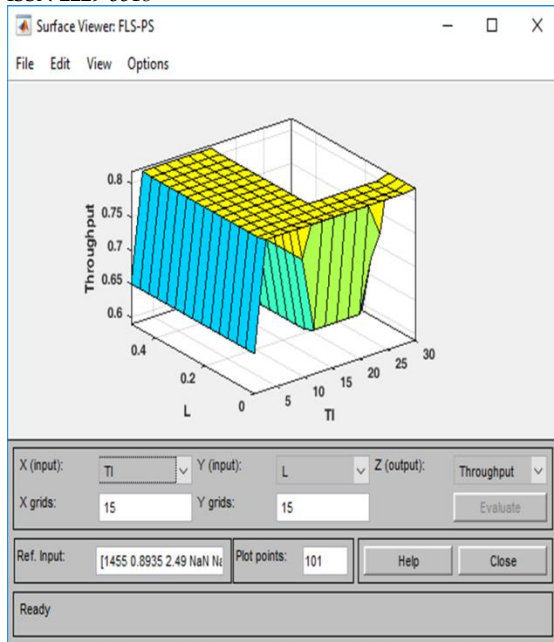


Figure 32: Interaction between latency and traffic intensity

### 5.3.4. Interaction between Latency and Transmitted Packet Length

The result in Figure.33 yield similar performance as Figure.32, indicating a relationship between transmitted packet length and traffic intensity. Of course, the rate of transmitted packets aggregated over time could certainly result in high traffic intensities resulting in highly congested systems. As packets are delayed, system aggregation surpasses the recommended threshold, and if not put to check in practical systems, may result in low system performance

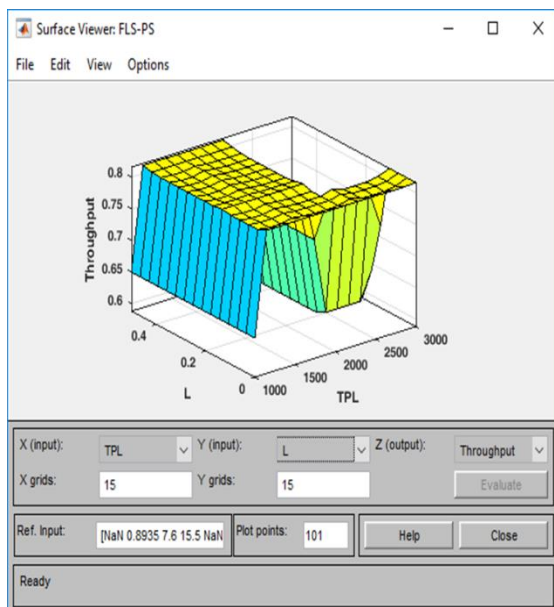


Figure.33. Interaction between latency and transmitted packet length.

### 5.3.5. Interaction between Transmitted Packet Length and Packet Arrival Rate

From the statistics of the existing system, we observed that the system throughput was quite low because of the inconsistent transmission packets. In order to improve the throughput, packet aggregation scheme can be considered such that before designing packet aggregation algorithm, it is worth investigating the effect of packet aggregation scheme on the performance of throughput and delay.

The packet size strongly influences the required number of packets and packet headers, and thereby the transmission overhead. Result obtained in Figure. 34. Indicates that throughput and packet length depends on the number of stations and packet arrival rates. Hence, the influence of transmitted packets irrespective of the length does not cause colossal degradation of the system throughput. In practical networks however, traffic is usually non-heavy and the delays in the source nodes do matter for the end-to-end delays experienced by applications.

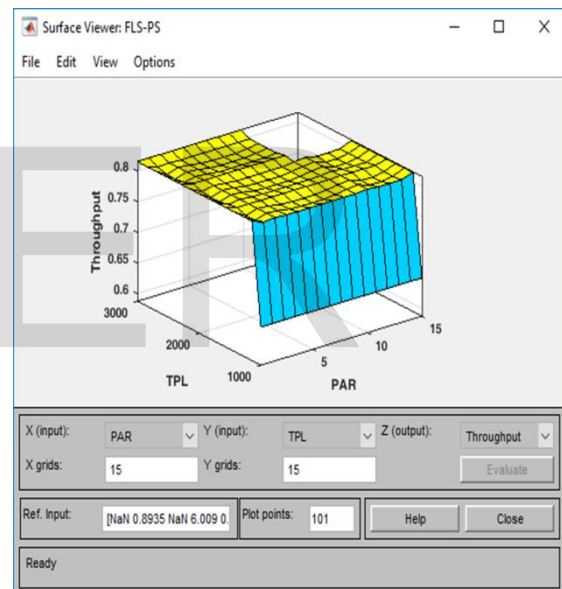


Figure.34: Interaction between transmitted packet length and packet arrival rate

## CONCLUSION

Optimal packet routing has been widely studied for various networks. This Thesis considered the optimization of packet switching using the throughput measure. An approximate reasoning approach adopting fuzzy logic was implemented to solve the packet-switching problem, such that the communication network can maximally utilize the allocated bandwidth. To obtain an adaptable system, real-life empirical data were obtained from an existing 3G network, where membership functions were constructed using averages of each parameter, and scaled for optimization purposes. Five parameters that directly influence packet-switching were considered for the purpose of predicting the optimum performance of the network. These parameters

were, transmitted packet length, packet loss, packet arrival rate, traffic intensity and latency.

## RECOMMENDATIONS

In modern communication networks, particularly in packet-switched networks, routing is an important process that has a significant impact on the network's performance. Ideal routing algorithm comprises finding the "optimal" path(s) between source and destination router, enabling high-speed data transmissions and avoiding packet losses. This Thesis therefore recommends the following:

(i) The existing system should adopt a robust routing strategy, since routing is most important in any communication network. The Fuzzy Logic system developed in this Thesis can offer seamless routing of packets, and in a controlled manner;

(ii) The objective of a routing strategy is essentially to minimize the mean delay of the packets in a network, subject to some reliability or capacity constraints. With the high capacity and intense communication nature of AKBC, the proposed system offers minimal latency and low packet losses, and is reliable for implementation and seamless transfer of packets in the existing network;

(iii) The concept of approximate reasoning implemented by the proposed system guaranteed the stability and reliability of the network. The existing system was not reliable in many respects. As can be observed in Chapter five, the throughput of the existing was below the recommended threshold, compared to the proposed system, which throughput was about 80%. Hence the proposed system demonstrates good performance and should be installed to improve the operations of the existing system functions.

## CONTRIBUTIONS TO KNOWLEDGE

This Thesis has made modest contributions to knowledge, as follows:

(i) The application of discrete solutions has often been adopted in analytical studies for ease of tractability. However, previously reported results obtained for heavy traffic may be misleading by indicating delay reductions. As demonstrated by our results, an intelligent approach using fuzzy logic is therefore important to maintain good system performance and improve the existing system. This Thesis has designed a robust system with useful parameters to optimize the functions of an existing system;

(ii) For practically relevant non-heavy traffic, there are some differences between the switch (network) settings, which become complicated for heavy traffic. Through approximate reasoning and numerical solutions, this Thesis has been able to prove the efficiency of fuzzy logic in packet switching. The proposed system has contributed to reducing packet losses caused by increased traffic, hence, improving the throughput performance, and most suitable for practical systems;

(iii) The proposed system guarantees improved quality of service. With the implementation of the underlying concept of linguistic variable definitions for

variables with uncertainties has been widely unrecognized—a fact which relates to the concept of precision. This Thesis offers precise/accurate solution to the packet switching problem, as opposed to non-linguistic specifications which produces unreliable solutions;

(iv). Adaptability of results is important for scalability purposes. This Thesis has presented an adaptable solution and can be scaled to related problems. The constructed membership functions can be easily tweaked to modify the network and improve the behaviour of the system.

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