Demand-driven, Concurrent Discrete Event Simulation

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To Catherine Smart, without whom this thesis would never have been started, and to David L. Lyle, without whom it would never have been completed.

Abstract

The simulation of complex systems can consume vast amounts of computing power. In common with other disciplines faced with complex systems, simulationists have approached the management of complexity from two angles: sub-system evaluation and level of abstraction. Sub-system evaluation attempts to determine the global behaviour by determining the local behaviour and then joining these behaviours together. Altering the level of abstraction tries to reduce the detail in the system in areas which are less critical to the model.

Data-driven evaluation, where the computation is sparked by the arrival of sufficient data, has been widely used as a basis for discrete event simulation. Demand-driven evaluation uses a different impetus for computation. It actively demands that data be sent in order for it to complete the processing. The demands that each processing unit issues, in turn, cause other processing units to become active. The repeated demand for finer and finer sub-solutions will eventually be satisfied which results, in turn, with the solution of the original demand. Demand-driven evaluation provides a coherent approach to the problem of simulating large systems at different levels of abstraction, at a cost comparable to data-driven evaluation. A model for both data- and demand-driven evaluation is described which captures the total communication and computation load for each node in the system.

Models are provided for the upper-bound of processor and communication usage. The runtime dynamics of data and demand-driven systems are investigated with particular emphasis on the relation between the costs of generating and transmitting an event.

Demand-driven discrete event simulation, using time intervals, is able to provide a platform with dynamic communication between the nodes, local control of processing, efficiently uses processor power, and is conservative. If the structure being simulated is free from deadlock, then the simulation will be also. The

client-server approach means that the evaluation is easy to distribute over available processors. The use of a calendar and time intervals means that the system is able to automatically identify, and exploit, both structural and temporal parallelism in the underlynig system.

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Declaration

I declare that this thesis was composed by myself and that the work contained therein is my own, except where explicitly stated otherwise in the text.

(*Colin Smart*)

Table of Contents

List of Figures

Chapter 1

Introduction

The problem of efficiently executing regular, parallel programs has been much studied, and machines such as the Connection Machine[41] were designed to facilitate such computation. Such early parallel machines were designed for parallel computation from the outset. Recently there has been a change of focus, away from monolithic systems, towards utilising a networks of workstations where the parallelism is supported more by the operating system and less by dedicated hardware. Examples of such systems are the SETI@Home[50] and Beowulf[86] projects.

The area of irregular computations, however, has been less extensively examined. Irregular computations are characterised by an execution pattern which cannot be predicted in advance and which is very sensitive to the input data. Parallel discrete event simulation is one such irregular computation and is used throughout to illustrate the methods employed.

1.1 The structure of the thesis

Chapter 1: Introduction. This chapter introduces distributed discrete-event simulation as a means to explore irregular computation and, after a review of the major approaches to time synchronisation in such systems, proposes a new method that addresses an aspect of efficiency which has been overlooked by the other approaches.

Chapter 2: Background. This chapter steps back from simulation and looks at the more generic problems of the production and synchronisation of data

in distributed systems, and how it relates to the desirable features of a dynamic communications topology, freedom from deadlock, local control and efficient use of resources.

Chapter 3: Demand-driven Simulation. This chapter discusses some of the costs and benefits associated with demand-driven simulation. The costs are resource consumption, be they bandwidth, processor or time. It provides arguments in mitigation of a number of the costs involved as well as strategies to reduce the overall cost of simulating a system.

Chapter 4: Performance models. This chapter first describes, in detail, the behaviour of Chandry-Misra-Bryant, ELSA and demand-driven systems. After providing background definitions, models are derived which express the upper-bound of the gross computation and communication behaviour of those systems.

Chapter 5: Experimental results. This chapter uses a number of different circuits to examine the dynamic nature of the simulation and, in particular, to focus on the parallelism and performance which is available as the computing resource increases.

Chapter 6: Summary and Conclusions. This chapter summarises our work, provides some discussion of our conclusions and gives some directions for further work in the area of demand-driven systems.

1.2 What is Simulation?

Computer simulation involves the construction of a mathematical model of a system in which mathematical symbols and equations are used to represent the relationships between objects in the system. The calculations indicated by the model's equations are then performed repeatedly, using a computer with time incremented discretely, to represent the passage of real-world time. The computer simulation indicates the behaviour of the mathematical model and from this is inferred the behaviour of the modelled system.

Computer simulation is currently used in a wide range of applications, especially in engineering and the physical sciences, where systems are expensive or difficult to analyse. Much of what is known about many safety critical applications is derived from computer simulation; for example, if testing of the real world system under extreme conditions would involve excessive risks, then simulation must be used to determine the system's likely behaviour. Similarly, the likely performance of a new system is often assessed from simulation studies. This is particularly so when, for safety reasons, a system cannot be allowed to 'go live' in an untested configuration, or when it is impractical to experiment with the environment with which the system interacts.

Clearly, the integrity of the computer simulation is of critical importance; the simulation must be designed with care, so that the results obtained are valid, accurate and useful.

Most systems may be defined as a collection of elements which inherently execute concurrently and interact one with another to achieve some global function. For example, the human heart, lungs and bloodstream form a physiological system whose purpose is to provide oxygen for the body; each component exists and operates largely autonomously, yet the overall function is achieved by the interaction of the components. By analogy, any model should include whatever concurrency and inter-process interactions exist in the real-world system, and the simulation should be able to handle that concurrency and inter-process interaction.

The ready availability of low-cost parallel processing elements makes it increasingly attractive to use true parallel processing and true process interaction in simulation. A number of specific problem domains have been explored and a variety of systems have been reported (a few of these systems are examined in detail below). These reports have shown that complex systems can be modelled easily and economically, keeping a close relationship between the model and the real-world system, and without compromising the natural concurrent nature of the real-world system. In addition, the use of parallel computers can lead to substantial performance gains.

There are two classes of model available: continuous and discrete. A continuous model is used where the system varies continually with time. A discrete simulation is used when we are more concerned with the transitions from state to state than with the times at which they occur. We shall look only at discrete models.

1.3 Types of Discrete Simulation

An *event* is an action which can occur within the system being simulated.

In a discrete simulation the state of the system is assumed to remain constant between events. By making the interval between events smaller and smaller, an approximation of a continuous system can be achieved, though there will always be inaccuracies.

1.3.1 Time Advance

The method for advancing time in a discrete simulation system can be used to partition the methods into two classes:

- Time-driven simulation. This method is also known as compiled mode simulation. In this system, the continuous flow of time is modelled as a succession of equally spaced steps. The entire system is evaluated for each of those steps. A disadvantage of this method is the inherent assumption that the state of the system at time $t+\delta t$ can be determined by some function of the state at time t and the inputs to the system at time $t+\delta t$. This method also fails to record changes to the system in the interval $(t, t+\delta t)$. It does, however, have the advantage that no scheduling is necessary (as the whole system is evaluated every δt). Also, it is relatively simple to implement on parallel or distributed machines as there is no synchronisation required between the components of the system to impede the execution.

- Event driven simulation. If we consider the system to be simulated as a number of elements, each of which maintains a local state which, in turn, is used as the input state to a number of other elements, then an event

driven simulation can be employed. The number of elements whose inputs change at any given time is generally quite small and much of the execution time in a time driven simulation is wasted, either recalculating an output whose inputs have not changed, or in checking to see what has changed. An alternative approach is to mark each change in state with the time at which that change takes effect. The simulator thus knows what, and when, states change. For some systems, the overhead in maintaining this extra state information makes the event driven system perform poorly compared with time-driven systems although it can perform better if the state changes are rare (either in time or space).

A timing model is used to mimic the time taken by a element to determine the new output value when one or more input values change. A number of different timing models are available.

- Unit delay assumes that every change of an input state requires exactly one time unit before its effect appears as an output state. It is worth noting that a change in the input state does imply a change in the output state. This is the only timing model available to time-driven simulation.

- Fixed delay assigns individual delays to each element and keeps these delays constant throughout the entire simulation. This can be used to mimic the granularity (or response time) of the element in question. Should the time taken to process a change in state depend on the specific transition being experienced by the element (from old input state to new input state) then multiple fixed delays can be applied.

- Variable delay provides a more flexible way to simulate elements. With this type, the value of the delay changes to reflect the state of the system. For example, a car waiting to cross a train track will have a delay which varies with the speed and length of the train; values which may be data dependent.

1.4 Classical Discrete Event Simulation

In a classical discrete event simulation system, a queue holds an ordered list of event-time pairs. The list is ordered on the time component of the pair. In effect, the pair dictates what happens and when it happens.

Each event can cause a number of other events, including itself, to be scheduled in the future. Some systems permit events to be scheduled at the current time, while others expressly forbid such scheduling in order to ensure the progress of time. No event can cause an event to be scheduled in the past. A simulation system, then, consists, in the abstract, of a single queue which holds the scheduled events in time order. Events with the same time-stamp are evaluated in an order determined by a resolution strategy. This strategy can be as simple as first-come first-served. In some models, the existence of two conflicting events, such as "increase heat" and "decrease heat", scheduled for the same time is an error condition which halts the simulation.

The simulation proceeds by executing the event at the front of the queue (the event with the lowest time-stamp) and inserting into the queue any resulting events. This continues until either a preset time or condition is reached, or the queue becomes empty.

Early attempts at parallelising the simulation were still based on the single queue model of the sequential method[27]. It was thought that, as there could be a number of events in the queues with the same time-stamp, a performance gain could be achieved by executing all such events on separate processors. While this did improve performance, such systems had a number of drawbacks, the most notable being that the single queue proved to be a bottleneck in the system. While one processor was executing the last of the events with the current time-stamp, the rest of the system had to wait until it had finished before issuing events with a higher time-stamp. If one event scheduled another event with the same time-stamp, then the system had to process these sequentially, with the resultant loss of parallel performance.

This was confirmed by Agrawal[2] and others. Work then began on a number of more complex queuing models which eventually resulted in the Chandy-

Misra[19] or Bryant[13] systems, which will be described in Section 1.5.

1.4.1 Shared-memory multiprocessors

There have been many attempts to apply parallel computers to discrete-event simulation. These may be divided into two main approaches, distributed simulation and concurrent simulation. Distributed simulation relies on a spatial decomposition and partitions the simulation model into components that can be executed on different processors. Concurrent simulation is based on a temporal decomposition.

While this thesis concentrates on distributed simulation, some developments in shared-memory concurrent simulation[23, 90] are worthy of mention. Many of the performance-degrading obstacles found in distributed memory simulations, such as communication delay, null messages, and the high cost of deadlock detection and recovery, can be reduced. Near ideal speed-up for several queuing network simulation models using shared-memory distributed simulation has been reported by Wagner and Lazowska[90, 91].

Hoeger and Jones[42] have integrated the two distributed and concurrent approaches. They have produced a distributed simulator with concurrency added to each model component. This was done in a shared-memory environment and both approaches were unified to an event-centered view. They partitioned the global event queue of the concurrent simulator and provided each model component in the distributed simulator with a local concurrent event queue which allowed them to add concurrency to each model component.

1.5 Distributed Simulation

The field of distributed simulation has received a great deal of interest and numerous methods have been developed to maintain a sufficiently accurate view of time across a collection of processing elements. In this section we shall start by providing a brief overview of distributed simulation and then follow with a survey of the different approaches that have been taken to address the issues raised by successive systems.

In distributed simulation, the physical system is usually modelled as a set of spatially separated physical processes that interact at discrete time instants. The distributed simulation approach maps each physical process onto a logical process (LP) of the simulation engine. Interaction between physical processes is handled via time-stamped messages, exchanged between the corresponding logical processes. Each LP maintains its own local clock — often referred to as Local Virtual Time (LVT) — and a local event queue holding messages in time order. A synchronisation protocol has to be provided and executed by each logical process in order to preserve the dependency between events in this asynchronous environment. In the simulation engine, the logical processes are mapped to processors; the communication links are embedded in the underlying inter-processor communication network. This provides a natural means, not only for exploiting parallelism, but also for maintaining the modularity of the simulation.

Two different styles of synchronisation have, until recently, further divided distributed simulation into two classes; conservative and optimistic.

1.5.1 Conservative Mechanisms

The essential basis of distributed simulation was first presented by Chandy and Misra[19], and independently by Bryant[13]. Such systems are sometimes referred to as CMB (Chandy, Misra, Bryant) systems.

In CMB systems, the causality of events across all the LPs is preserved by sending time-stamped event messages (<event@t>); the time-stamp is a copy of the LVT of the sending LP. A conservative logical process is allowed to process *safe* events only. A safe event is one which has a time-stamp in advance of the LVT of the receiving LP, but less than (or equal) to the time-stamps on all other messages which the LP will receive. All events must be processed in chronological order. This guarantees that the output stream of a LP is in chronological order. A communication system preserving the order of messages sent from one LP to another (FIFO) is sufficient to ensure that no out of chronological order messages will ever arrive at receiving LP. A conservative system can thus be seen as a set of all LPs together with a set of directed, reliable, FIFO communication channels

that constitute a graph of logical processes. It is important to note that this graph has a static topology.

The communication interface of a logical process maintains an input buffer and a clock for each channel pointing to that LP. The buffer stores every message arriving through a channel in FIFO order and that channel's clock is set to the time-stamp of the earliest unprocessed message (the one at the head of that channel's buffer). Initially the value of every channel clock is set to zero.

The local virtual time is the minimum of the channel clocks. This gives the time horizon, up to which it is safe to process events. It is safe because, given the FIFO links and a fixed topology, it is not possible for any LP in the system to send a message down a channel with a time-stamp less than already sent and no LP can send a message without having started at a LVT of 0.

The event (or events) with a time-stamp equal to the LVT are processed and removed from the input buffer and any resultant events dispatched. Given that there are now no messages left with a time-stamp equal to LVT the LP can perform one of two actions. If there is a message on all of the input arcs then the LP can increase its LVT to the new minimum and repeat, or it must wait until all the channels have messages before repeating. This "blocking until safe" policy leads to two problems: deadlock and memory overflow as shown in Figure 1.1. Each LP is waiting for a message to arrive from a LP which is itself blocked (deadlock). Also, each process which is blocked is receiving messages from non-blocked LP which are being queued and left unprocessed in their respective input buffers. These input buffers can grow unpredictably and thus cause memory overflow. This is possible even in the absence of deadlock. Several methods have been proposed to overcome the vulnerability of CMB to deadlock, these fall into two principal categories: deadlock avoidance and deadlock detection/recovery.

1.5.1.1 Deadlock Avoidance

Deadlock, such as that in Figure 1.1, can be prevented by modifying the communication protocol so that *null messages*[60] (messages of the form <null@t>, where null is an event with no effect) can be sent. A null message is not related

to the simulated model and serves only as a synchronisation method. It is sent on every output channel as a statement that that LP has reached a certain value of LVT and thus will never send out a message with a time-stamp less than t. A null message is sent to every target LP for which the sending LP did not generate any other message. The effect is to notify every target LP of the sending LP's new LVT. The receiving LP can use this information to increase the channel clock on the corresponding link and thus permit other events to be processed.

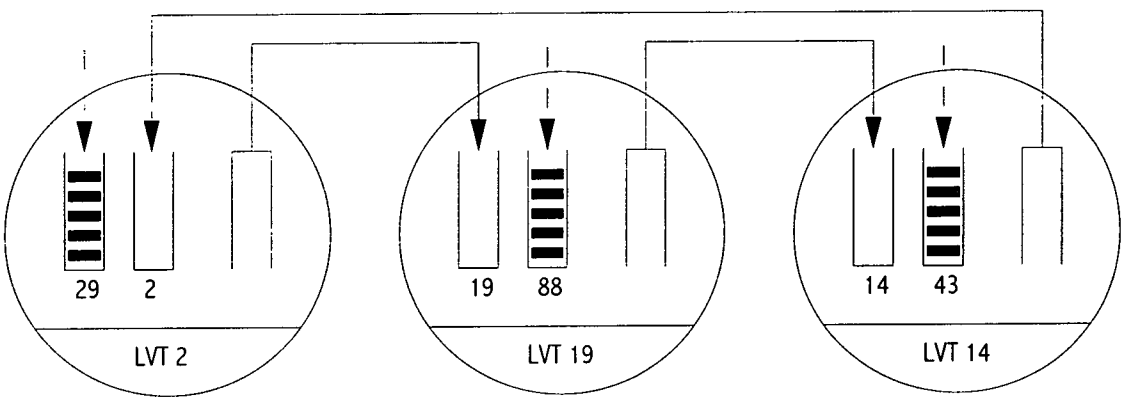


Figure 1.1: Deadlock and Memory overflow. The number beneath each channel denotes the time-stamp of the earliest unprocessed message (the channel clock).

In Figure 1.1, after the LP in the middle had sent <null@19> to the neighbouring LPs, both of them could increase their LVT to 19 and in turn issue new event messages to other LPs. The null message protocol can be guaranteed to be deadlock free as long as there are no closed cycles of channels, for which a message traversing this cycle cannot increase its time-stamp. This implies that simulation models cannot be simulated using CMB with null messages, if they cannot be decomposed into LP such that for every directed channel cycle there is at least one LP to put a non-zero time increment on traversing messages.

Although the protocol is straightforward to implement, it can put a greatly increased burden on the communication network (as a result of the null messages) and also reduce the performance of the simulation, as each null message needs to be processed. Optimisations on the protocol to reduce the frequency, or number, of null messages have been proposed[60]. An approach whereby additional information is carried with the null message (the so-called carrier-null message

protocol[17]) will be looked at in Section 1.5.1.2.

One remaining problem with trying to improve the performance of conservative logical processes is determining when it is safe to process an event. The degree to which LPs can *look ahead* and predict future events can play a critical role in the safety verification, and thus the performance, of conservative LP simulations. In Figure 1.1, if the LP with LVT of 19 knew that processing the next event will increment the LVT to 22 then it could send a null message <null@22> (a *look-ahead* of 3) to improve the LVT of the receivers.

Look-ahead must come directly from the underlying simulation model and enhances the prediction of future events; the ability to exploit look-ahead was first shown by Nicol[66] for FCFS queuing network simulations.

1.5.1.2 Carrier-Null Message Protocol

As mentioned in the previous section, it is possible to augment the null message with other information to help overcome some of the inefficiencies of the null message protocol. Consider the system shown in Figure 1.2. The source creates an event every 50 virtual time units; the join, split and pass units each take 2 virtual time units to handle the event. After the first event is released by the source, all LP except the source are blocked and start to propagate local look-ahead via null messages. After 4 null messages (join to pass, pass to split, split to join and split to sink) each of those LP has advanced their local time by 2 virtual time units. It will take a further 96 null messages (100 in all) before the initial source event can be processed and then another 100 null messages before the second source event can be processed, and so on. The impact of look-ahead is easily seen in this example; the smaller the look-ahead on the successor LPs, then the more null messages that will have to be sent to advance the virtual time, resulting in a higher communication load and thus a poorer performance. In a study by Leung and others[51] it was shown that cycles in the communication network of a conservative CMB system can remove almost all the speedup from the system.

The carrier-null message protocol[17] aims to reduce the number of null mes-

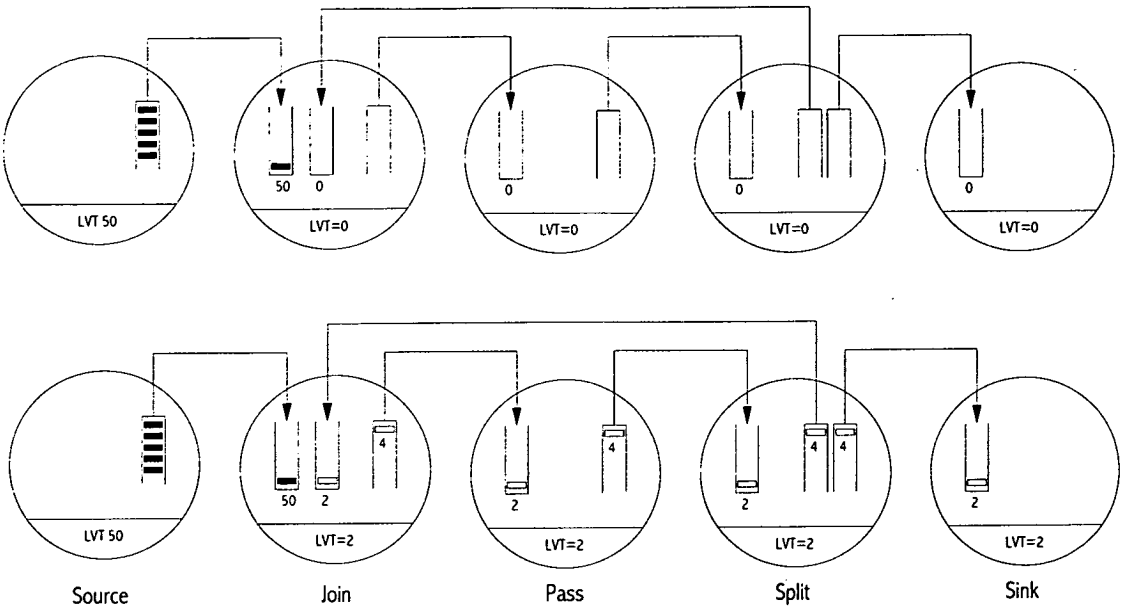


Figure 1.2: Motivation for Carrier-Null Message Protocol

sages sent by augmenting the message with a number of other parameters. If the join process in Figure 1.2 could somehow know that it is waiting on itself, it could safely process the source event (t=50). To do this, the LP_{join} needs some global information. To satisfy this need for global information, without having a centralised controller, the carrier-null protocol employs an additional null message of type <c0, t, R, la.inf>, where c0 is an identification as a carrier null message, t is the time-stamp, R contains an ordered list of the logical processes through which the message has been routed and la.inf is look-ahead information. Once LP_{join} has received a carrier null message with itself as the source and sink of the route in R, it can be sure that (in this example) it will not receive an event message via that path unless it itself had sent an event message along that path. It can, therefore, after receiving the first carrier null message, process the source event and thus increment its own (and the other LPs in route R) LVT.

In the more general case, where there may be more than one "source-like" LP entering event messages into the dependency loop, the above arguments are not sufficient as more information is needed than just the route taken. The earliest possible time of next event message that would break the cyclic dependency is also needed. This is carried in the field la.inf in the carrier null message.

Even with carrier null messages, the CMB system can still produce many null messages. An approach by Preiss and others[72] attempts to reduce null message propagation by recognising when a null message has become superseded (or stale). Suppose that a LP has sent a stream of null messages to another LP. For example, this might occur when the originating LP has more than one input channel. Each of these null messages will have an increased time-stamp. The null messages will be queued at the input buffer until being processed. Should a null message with time-stamp t arrive in the buffer and find another null message with a time-stamp $s < t$ then there is no point having the receiving LP process the earlier null message as it it now redundant and can be annihilated. This was generalised further to say that *any* message from the same source which finds a null message with a smaller time-stamp may annihilate that null message. This optimisation depends on the respective rate of production and consumption of null messages and may, in the case where the LP is a greedy consumer, produce no performance improvement whatsoever.

A later study, by Teo and Tay[88], of the conservative simulation of a multi-stage interconnection network uses a similar "flushing" method to that proposed by Preiss[72]. In the example used by Teo and Tay, the amount of null message overhead was reduced from exponential to linear in the number of elements in the system. This has important repercussions on the performance of the system as Soule[82] notes that, in parallel event-driven simulation of logic circuits, 50% to 80% of the execution time is spent in the deadlock detection and recovery phases.

1.5.1.3 Deadlock detection and recovery

An alternative to the null message approach was also proposed by Chandy and Misra[19], which allowed deadlocks to occur but provided a method to detect them and recover. Their algorithm has two phases: the first (a parallel phase), in which the simulation runs until it deadlocks, and the second (an interface phase), that starts a computation which results in at least one LP being able to advance its LVT. They prove that, in every parallel phase, at least one event will be processed, generating at least one event message which will also be propagated

before the next deadlock. Their algorithm assumes a *central controller*, which violates a central tenet of distributed computing. This was later removed and replaced with a distributed deadlock detection algorithm[20].

Misra[60] proposes an alternative approach in which a special message (called a *marker*) circulates through the network of channels to detect and resolve deadlock. A cyclic path for traversing all the channels is precomputed and all LPs are initially coloured *white*. A LP that receives the marker turns white and forwards it along the path in *finite* time. Once a LP has forwarded the marker, should it either send or receive an event, then it turns *red*. Deadlock is detected by the marker if the last N LP visited were all *white*. If the marker also carries the next event times of the visited (white) LPs then it will know, once it has detected deadlock, the smallest next event time as well as the LP in which this is supposed to occur. To recover from deadlock, this LP is invoked to process its earliest event.

The *time-of-next-event algorithm* proposed by Groselj and Tropper[38] assumes more than one LP mapped to a single physical processor and computes the lower bound of the time-stamps of the event messages expected to arrive next at all empty links on the LPs located at the processor. It thus helps to unblock LPs within one processor but does nothing to prevent deadlocks across processors.

An optimisation has been adopted by Soule and Gupta[83]. Their work is specific to logic simulation and centres on manipulating the order in which nodes are evaluated to reduce the potential for deadlock. In some cases, all deadlock has been removed.

1.5.1.4 Summary of conservative methods

The principle of conservative operation is that causality violations are strictly avoided; only "safe" events are processed. The synchronisation method is process blocking, which can cause deadlock. This is inherent in the protocol and not a resource contention problem. Deadlock prevention protocols based on null messages are liable to place a severe communication overhead on the system. Deadlock detection and prevention algorithms mainly depend on a centralised

controller, though other methods are available. The parallelism available within a CMB system is purely *structural* and rarely fully exploited as, if causality violations are possible, even if rare, the protocol behaves overly pessimistically as it waits until it is not possible for a violation to occur. CMB performs well as long as all channels are equally utilised. Should a channel not have a new event message, because the state has not changed, then either it will need to send null messages or become involved in a deadlock detection and recovery process. A large dispersion of events in either space or time does not degrade performance. This is because a conservative LP is only concerned with the earliest message from those LPs that are directly connected to it. The potential zone of influence of a LP is small and thus it is relatively insulated from the rest of the simulation system.

There is no explicit computation of a global virtual time (GVT) which, as we will see in Section 1.5.5, is needed to manage memory in optimistic systems. The global virtual time is the time before which no events can occur.

A conservative system can cope with simulation models having "arbitrarily" large state spaces and is straightforward to implement using only simple control and data structures, though it does require that the communication channels are FIFO and that events are processed in the order of their arrival (which will be, under the strictures of the protocol, in chronological order). The LP interconnection topology must be static.

While no general performance statement is possible owing to the many different systems, implementations and architectures, the performance of a CMB system relies mainly on its deadlock management strategy. The computation and communication overhead per event is small on average and the protocol favours "fine grain" simulation models.

1.5.2 Optimistic Mechanisms

The "pessimistic" causality constraint of the conservative system strictly prevents any *out of order* execution of events. In contrast, optimistic LP simulation strategies allow causality errors to occur and provide a method whereby the system can

recover from such violations. In order to avoid the blocking and safe-to-progress determination which hinder the performance of conservative systems, optimistic processes evaluate events (and hence advance LVT) as far into the future as possible. This is done with no regard for causality errors and there is no guarantee that an event will not arrive in the local past.

1.5.2.1 Time Warp

The initial work in optimistic simulation was by Jefferson and Sowizral[45, 48] with the definition of the Time Warp (TW) mechanism which, like the Chandy-Misra-Bryant protocol, uses messages for synchronisation. The Time Warp mechanism restores consistency with the local causality constraints[34] through the use of a *rollback* mechanism. If an event arrives with a time-stamp in the local past, i.e. out of chronological order (these messages are sometimes referred to as *straggler messages*), then the TW scheme rolls back time to the most recently saved state in the LP history which is consistent with the time-stamp on the new message and restarts the simulation from that point.

Rollback requires a record of the history of the LP so that it can return to a point in its past and correct the causality error. This mean a record not only of internal state changes, but also of the contents of input and output queues. For reasons which we will cover later, the record of the LP's communications history must be done in chronological order.

Since the arrival of event messages in increasing time-stamp order cannot be guaranteed, two different kinds of messages are required to implement the communications protocol. The first is the usual CMB style message but with an added '+' field ($m^+ =< ee@t, + >$), where again ee is the event and t is a copy of the sender's LVT. Subsequently we will refer to this type of message as a *positive* message. To balance positive messages, we also have *negative* messages (or anti-messages) of the form ($m^- =< ee@t, - >$). These negative messages are transmitted to a LP to request the annihilation of the prematurely sent positive message containing *ee*. This would occur when the sending LP discovered that the value of *ee* was computed based on a causally erroneous state.

The basic architecture of an optimistic LP is similar to that for a conservative LP. Again messages are transmitted through a communications system but they are not required to arrive in the order that they were sent and this relaxes the hardware requirements. Also it is not necessary to separate the input streams, so a single input queue is sufficient (as long as the sending LP can be identified from the message). The communication history must be stored, as must the internal state.

An optimistic LP works in four phases: input synchronisation to other LPs, local event processing, the propagation of external effects and the global confirmation of locally simulated events. The event processing, and propagation of external effects, are almost the same phases as those contained within a conservative system. The input synchronisation (rollback and annihilation) and confirmation are the key elements in an optimistic LP simulation.

1.5.3 Rollback and associated Annihilation Methods

The rollback mechanism relates the incoming message with the current state of the LP to determine the appropriate action. There are three possible variables to consider; the type of the arriving message (m^+, m^-), the relation of the time-stamp to the LVT (time-stamp $\geq$ LVT, time-stamp $<$ LVT) and whether a dual message exists (a m^+ for a m^- or a m^- for a m^+). The appropriate action is outlined in Tables 1.1 and 1.2.

	Arriving message is of type: m^+	
time-stamp $\geq$ LVT (in the local future)	if dual m^- exists annihilate dual m^-	if dual m^- does not exist chronologically insert (m^+,IQ)
time-stamp $<$ LVT (in the local past)	if dual m^- exists annihilate dual m^-	if dual m^- does not exist rollback then chronologically insert (m^+,IQ)

Table 1.1: Appropriate actions on receiving an incoming positive message in a Time Warp based protocol

Events which arrive in the *local* future are unproblematic as they have yet to be processed and, as such, cannot have had an effect outwith the local environment. So, should a positive message arrive it will either a) cancel out an existing negative

	Arriving message is of type: m^-	
time-stamp $\geq$ LVT (in the local future)	if dual m^+ exists annihilate dual m^+	if dual m^+ does not exist chronologically insert (m^-,IQ)
time-stamp < LVT (in the local past)	if dual m^+ exists rollback then annihilate dual m^+	if dual m^+ does not exist chronologically insert (m^-,IQ)

Table 1.2: Appropriate actions on receiving an incoming negative message in a Time Warp based protocol

message or b) should no related negative message exist, it will be inserted in the queue in time-stamp order. The arrival of a negative message will be treated similarly in that it will either a) cancel out an existing positive message or b) should no related positive message exist, it will be inserted in the queue in time-stamp order. The effect of such actions is to ensure that an erroneous positive message is cancelled (even if it arrives after the cancelling negative message). As all processing so far discussed takes place in the local future there is no need to involve any other LPs as they could not have received any output from this LP in its local future that has not already been corrected. Situations in the lower row, where the event arrives in the LPs local past, *may* involve other LPs if a causality error has occurred.

In the lower row, the arriving message is in the local past of the LP. That means that the LP has sent, to other LPs, data which *may* be erroneous. The two simple cases are a positive message arrives and there is a corresponding negative message, or a negative message arrives and there is no corresponding positive message. In the former case, the negative message is annihilated. In the latter case the negative message is inserted into the input queue in chronological order. The remaining two cases need rollback to be implemented.

In the case of a positive message arriving in the local past with no corresponding negative message then the system must rollback to a point before the time-stamp of the arriving message, insert the new message into the input queue in chronological order and then restart the simulation. How the simulation can be rolled back is dealt with below. In the case of a negative message arriving, that has a corresponding positive message (which has already been processed),

the system must roll back to point before the time-stamp of the arriving message, annihilate the associated positive message, and then restart the simulation from that point.

As can be seen, the rollback mechanism requires a periodic saving of the state of the LP. This allows the LP to rewind to some state before the causality error occurred and then to continue processing past the *now corrected* error. It is also necessary to maintain a log of all outgoing messages in order to undo events which have been propagated to external LPs. Observe from the table that anti-messages can also cause rollback and, as such, can cause *rollback chains* in which one LP, in rolling back, causes other LPs to rollback. It is even possible for recursive rollback to occur should a LP in a cycle start to rollback. The protocol guarantees that *any* rollback chain will eventually terminate whatever its length or recursive depth. Such a rollback chain can consume significant memory and communication resources.

1.5.3.1 Aggressive Cancellation

The original Time Warp protocol described, in part, above used aggressive cancellation. Using this form of cancellation whenever a *straggler message* (a message with a time-stamp in the local past) arrives, anti-messages are sent immediately to cancel all *potentially* incorrect messages. The aim of this was to reduce the number of potentially erroneous messages being processed by external LPs which may, in turn, force them to roll back.

1.5.3.2 Lazy Cancellation

A different cancellation policy was proposed by Gafni[36], which he termed *Lazy cancellation.* In this alternate policy, the system does not send anti-messages immediately upon the receipt of a straggler message. Instead the system delays the propagation until the LVT has, after rollback, reached the time-stamp on the straggler message and the system produces a different output message from that originally sent at that time-stamp. In this case the earlier message which was sent has been shown to be incorrect and needs to be cancelled. If the resimulation resulted in the same message being generated as had originally been sent then no

cancellation is necessary. Lazy cancellation thus avoids unnecessary cancellation of correct messages but does have the overhead of additional memory and book-keeping (potential anti-messages must be maintained in the output queue). It also delays the cancellation of incorrect messages, which may result in more rollbacks being needed downstream of the causality error.

The idea of lazy cancellation can be expanded, using the look-ahead value (la) first mentioned in Section 1.5.1.1, to reduce the number of rollbacks that are needed to maintain causality. If a straggler message $ts(m^+)$ <LVT is received then there is no need to send anti-messages for any message with a time-stamp less then $ts(m^+) + la$. Also, if $ts(m^+) + la \geq$LVT then rollback does not need to be invoked.

Jefferson[47] has shown that Time Warp with lazy cancellation can outperform the simulation's critical path. This is possible because calculations based on the assumed state of the system, which was later confirmed to be correct, would have propagated further through the system than they would have done under either conservative or aggressive cancellation strategies. This has the effect getting the correct value to the input of an element before it has been confirmed. This effect was termed "supercritical speedup". Aggressive cancellation does not have this potential as rolled back computations are discarded immediately. A comparison of the performance of the two is, however, related to the simulation model. It has been shown by Reiher et al.[75] that lazy cancellation can arbitrarily outperform aggressive cancellation and *vice versa*. While their study used synthetic extreme cases to highlight the strengths and weaknesses of each protocol, the empirical evidence is reported "slightly" in favour of lazy simulation for certain applications[36, 34].

Fujimoto[33] has adapted distributed simulation to the shared-memory multiprocessor environment, and also utilises shared memory to optimise the message cancellation process. The handling of roll-back can be a major overhead in a simulation and, as such, work has been done on providing hardware support for this operation[35].

1.5.3.3 Breaking or Preventing Rollback Chains

A number of other techniques, beside lazy cancellation, have been used to limit the number of rollbacks in a system. One approach, which was based on the carrier-null approach discussed earlier, was proposed by Prakash and Subramanian[71]. They attached a limited amount of state information to messages to prevent recursive rollbacks. The attached state information allowed LPs to filter out those messages which were based on an assumed state of the system. These false positive messages would eventually be annihilated by chasing anti-messages which were currently in transit.

Madisetti, Walrand and Messerschmitt[56] proposed a protocol called *Wolf-calls*. In this protocol, events based on the assumed state of the system, are able to propagate to a limited set of LPs within a specified distance of the source LP. These *spheres of influence* are defined as the set of LPs which can be affected by an event in a certain time (respecting both communication and computation times). The effect is to limit the number of LPs which can be affected and thus limit the length of the rollback chain. Dickens and Reynolds[28] proposed a variation on this idea with the SRADS protocol in which, while allowing optimistic progression, the propagation of uncommitted events to external LPs is prohibited. This means that rollback is local and that rollback chains can never occur.

1.5.4 Memory management in Optimistic Systems

The discussion so far has assumed the availability of *sufficient* free memory to store internal and external history for pending rollbacks. Lin[54] argues that Time Warp *always* consumes more memory than sequential simulation and that limiting the memory imposes a performance decrease. Providing merely the minimum amount required causes such a decrease in performance that a memory/performance tradeoff becomes an important issue.

There are two ways of limiting the amount of memory used in an optimistic system: i) reduce the amount of optimism as occurs in the systems proposed by Madisetti and by Dickens or, ii) save the state of the system infrequently or incrementally. Neither system can guarantee that memory will not be exhausted

and so it is necessary for the protocol to recover memory no longer needed by the system. This *fossil collection* is used to reclaim the memory being used to store events and states which will never be needed by the system because the global virtual time has progressed beyond their time-stamp. The global virtual time is the minimum time-stamp on any unprocessed event in the system.

1.5.4.1 Incremental State Saving

Many models have large and complex internal states which have to be stored. With each processed event, some of the variables which comprise the state will change while others will remain unchanged. An improvement can be made by only saving the variables which have changed. This "incremental state saving" was first proposed by Bauer et al.[8]. The incremental state saving can also increase efficiency as less data needs to be written to the log. This optimisation does, however increase the complexity of a rollback, as the desired state has to be reconstructed from increments following back a path further into the past than is required by the rollback itself. Lin[52, 55] studies the optimal checkpointing interval (how often to save the state), explicitly considering the state saving and restoration costs. He produced an algorithm which, while increasing the rollback overhead, can reduce overall execution time.

1.5.5 Global Virtual Time (GVT) Computation

In the descriptions of optimistic systems so far we have assumed that a global virtual time (GVT) is available at any instant on any LP. This is needed for fossil collection and the simulation stopping criterion.

The GVT is either the minimum LVT of any LP or the minimum time-stamp on any unprocessed message, whichever is smaller. The GVT has certain useful properties:

- At any real time $\mathcal{T}$ the $GVT(\mathcal{T})$ represents the maximum lower bound to which any rollback could ever backdate any LVT.
- $GVT(\mathcal{T})$ is non decreasing over real time $\mathcal{T}$ and therefore can guarantee that the simulation will eventually progress by committing intermediate

simulation results.

- Any processed messages or states at time $\mathcal{T}$ which have a time-stamp $ts < GVT(\mathcal{T})$ are obsolete and can play no further part in the simulation.

The efficient calculation of GVT is therefore another important issue to make the Time Warp system useful. Frequent invocations of the GVT calculation can result in a severe performance bottleneck owning to the communications load it places upon the system. However, in terms of simulation time, infrequent invocation causes a build up of uncommitted events and threatens memory exhaustion due to fossil collection being delayed. The optimal interval for performing a global virtual time (GVT) calculation has been extensively studied[73, 69, 18, 52].

The computation of GVT($\mathcal{T}$) is time-consuming and complex. This is because, as you can see from the definition, to obtain it requires the processing of a "snapshot" of the system, including all messages in transit at that point. As such, in practice an approximation $\widehat{GVT}(\mathcal{T})$ is calculated instead.

1.5.5.1 GVT Computations using a Central Manger

Naïvely, $\widehat{GVT}(\mathcal{T})$ can be computed by a central manager broadcasting a request to all LPs for their current LVT and performing a *min-reduction* on the collected results. This solution does not provide an entirely satisfactory answer as i) messages in transit could potentially roll back a reported LVT and, ii) all reported LVT values were sent at different real times.

These problems can be addressed by acknowledging the message carrying the LVT and by considering the GVT estimate to be true at some point in a real time interval. Samadi proposed an algorithm[78] in which the central manager triggered a GVT calculation by sending a *GVT-start* message. Once all LPs have reported, the central manager calculates, and broadcasts, a new GVT and ends the GVT calculation phase. The "message-in-transit" problem is solved by acknowledging *every* message and reporting the minimum time-stamp of all *unacknowledged* messages as well as the local LVT. The algorithm was later improved upon by Lin and Lazowska[53]. In their protocol, every message has a sequence number and, upon the receipt of a control message, the smallest number in the

sequence not yet received is sent to the originating LP as an acknowledgement of all messages with a smaller sequence number. By knowing what messages are still in transit it is possible to compute a lower bound on their time-stamps.

Bellenot[9] places a balanced binary tree over the network of LPs for the calculation of GVT. This more efficient algorithm uses (for N LPs) less then 4N messages and $O(\log(N))$ time per GVT epoch. His system requires, in common with that of Samadi and Lin, a fault-free, FIFO, communications structure.

The **passive response** GVT algorithm of D'Souza et al.[29] can cope with faulty channels while, at the same time, relaxing the need for a FIFO communication structure and also addressing the issue of centralised control. The heart of the protocol is the idea that each LP can determine when to report new GVT information to the central manager. A key improvement in this algorithm is that LPs simulating along the critical path will report more frequently than others. Logical processes which are processing far in advance of the GVT are much less likely to have an effect on $\widehat{GVT}$. This means that the communication resources are targeted at those LPs most likely to advance GVT.

1.5.6 Time Buckets

The *Breathing Time Bucket* (BTB) protocol[84] attempts to address the instability in Time Warp performance caused by anti-messages. The BTB protocol is an optimistic windowing mechanism with a pessimistic message propagation policy. As such, anti-messages are never needed and rollback is contained within the local LP (as in SRADS[28]). BTB processes events in *time buckets* of different sizes. The size of the bucket is determined by the *event horizon.* Each bucket contains the maximum number of causally independent events which can be executed concurrently. The *local* event horizon is the minimum time-stamp on any *new* event scheduled as a consequence of the execution of an event in the current bucket in some LP. The *global* event horizon (GEH) is the minimum over all local event horizons and defines the lower time edge of the next event bucket. Events are executed optimistically but events are only propagated if the GEH is greater than or equal to their timestamp. Two methods have been proposed to

determine when the last event in the current bucket has been processed and the distribution/collection of event messages generated within that bucket can start.

1. *multiple asynchronous broadcasts* to exchange the local event horizons in order to determine locally the GEH

2. *a system wide non-blocking sync operation* can be issued by every LP as soon as it exceeds the local EH. This does not hinder the LP and it can continue to optimistically process events. Once the last LP has issued the non-blocking synch, all the LPs are interrupted and requested to send their event messages.

Neither of these methods has an efficient software implementation and so they may need hardware support to be viable. Also, BTB can only work efficiently if *sufficient* events are processed on average in each bucket.

Steinman proposed a protocol called *Breathing Time Warp*[85] which combines the features of Time Warp and BTB in an attempt to eliminate the shortcomings of the two protocols. The underlying assumption is that the probability of having to cancel a message increases with the distance between the GVT and the time-stamp of the message, i.e., messages near GVT are more likely to be correct but messages well in the future are less certain. The proposed protocol operates in two modes, a Time Warp mode and a BTB mode. Each cycle starts in the Time Warp mode sending up to M output messages aggressively in the hope that they will not need to be cancelled. If the LP needs to produce more messages optimistically then the LP switches to BTB mode in which these optimistic messages are not propagated. Should the event horizon be crossed in BTB mode then a GVT computation is triggered followed by fossil collection. If the GVT is improved then M is is adjusted accordingly.

1.5.7 Hybrid Mechanisms

Traditionally the mode of simulation has been common to all LPs in the system. Recently, there has been increased interest in permitting processes in the simulation to run with either conservative or optimistic synchronisation mechanisms

and to permit them to change their synchronisation mechanism dynamically in response to internal events[5, 6, 74].

Reynolds[76] was the first to propose a mixed mode simulation system. The first implementation was by McAffer[57]. The system is characterised by two variables:

- Degree of aggression – this non-negative value determines how far in advance of a safe state the LP can evaluate. A safe state is one for which all the inputs are known and which is not threatened by rollback. This determines how *locally* optimistic the LP can be.

- Degree of risk – this value determines how far in advance of a safe state the LP can propagate the results of its execution. It has a non-negative value, and is less than or equal to the degree of aggression. If the degree of aggression is greater than the degree of risk then the precomputed results are stored locally.

If the degree of aggression of a LP is zero (and thus the degree of risk must also be zero), then the LP is executing as a conservative LP. If the degree of aggression is greater than zero and the degree of risk is zero then the LP is locally optimistic but globally conservative as it will not propagate potentially erroneous values. This, in effect, defines the SRADS protocol of Reynolds[28] mentioned earlier.

Cases where the value of risk is non-zero are "true" optimistic LPs in that they will propagate possibly incorrect events and the recovery from any incorrect event will be distributed across a number of LPs.

When LPs are firing in different modes the interface between them becomes important, to ensure the correct operation of all LPs in a system. There are four cases to consider:

1. **Primary inputs,** which are the source of events being inserted into the simulation system, can be connected to nodes firing in any mode as only correct information will be placed on these inputs.

2. **Conservative → Optimistic** can be connected as the conservative LP will only produce events which are safe. For this case a LP with a zero degree of risk can be considered conservative as no unsafe events will be sent.

3. **Optimistic → Conservative** cannot be connected directly. The optimistic LP, with a non-zero risk, may produce events which are unsafe. As any node with a degree of aggression of zero cannot recover from incorrect information, it is necessary to ensure that only safe events are received. This can be achieved by placing a buffer LP, with a degree of aggression of infinity and a degree of risk of zero, between the two LPs.

4. **Primary outputs** must receive events from a safe source (a LP with a risk of zero). This ensures that only safe data is passed as a result of the simulation. Again, this can be achieved by preceding the output LP with a buffer LP as described above.

1.5.7.1 Coarse-grain hybrid systems

Avril and Tropper[7] proposed a hybrid system called Clustered Time Warp (CTW). It is an algorithm for the parallel simulation of discrete event models on a general purpose distributed memory architecture. CTW has its roots in the problem of distributed logic simulation. It is a hybrid algorithm which makes use of Time Warp between clusters of LPs and a sequential algorithm within the clusters. This results in a two level simulation system with Time Warp being used to synchronise LPs which are, in fact, conservative simulation systems.

They developed a family of three checkpointing algorithms for use with CTW, each of which occupies a different point in the spectrum of possible trade-offs between memory usage and execution time. Their results showed that one of the algorithms saved an average of 40% of the maximal memory consumed by Time Warp while the other two decreased the maximal usage by 15 and 22%, respectively. The latter two algorithms exhibited a speed comparable to Time Warp, while the first algorithm was 60% slower.

1.5.8 Summary of optimistic methods

In optimistic simulation, causality violations do occur but are eventually detected and corrected. The synchronisation (and correction of erroneous events) is by a *rollback* of simulation time. Remote annihilation methods are liable to severe communication overhead. Rollbacks can cascade and, though they will eventually terminate, can reduce performance and increase memory usage.

The structural parallelism in the model can be fully exploited. The Time Warp system performs well if average LVT progression is "balanced" across all LPs though space-time dispersion of events can degrade performance.

Optimistic systems rely on explicit GVT calculation which can be hard to compute. Centralised GVT calculation systems are liable to communication bottlenecks if no special hardware support is given. Distributed GVT systems impose a high communication overhead and appear to be less effective.

Logical processes need to store state in order to recover from causality violations. This state consists of the internal state of the LP as well as its input and output event queues. The computation and memory cost of saving and restoring state can be large though incremental state saving can reduce this. Optimistic systems perform best when the state space, and the amount of memory needed to express the state, is small. Fossil collection requires frequent and efficient GVT calculation and complex memory management schemes are necessary to prevent memory exhaustion.

Messages can be delivered out of chronological order but must be executed in time-stamp order. Messages arrive in a single input queue and there is no need for a static communication topology.

The performance of the system relies mainly on controlling the optimism of LPs and on the strategy to manage memory consumption. The computational and communication overhead per event is high on average and thus the protocol favours "large grain" simulation models.

1.6 A desirable simulation system

A brief summary of the key features in both conservative and optimistic systems is given in Table 1.3. This table also includes the characteristics of a desirable system, namely a dynamic communications topology, local (distributed) control, efficient memory usage and freedom from artificially imposed deadlock. As such it should have some of the characteristics of both the optimistic and conservative systems.

	Communications Topology	Local or Global Control	Memory Efficient	Deadlock Free
Conservative	Static	Local	Yes	No
Optimistic	Dynamic	Global or Local	No	Yes
Desired	Dynamic	Local	Yes	Yes

Table 1.3: A brief summary of the features of conservative and optimistic simulation systems and the desired attributes of an ideal system

The justification of the desired features is as follows:

Dynamic communications: Certain domains of interest are, by their nature, static. For example, the logic simulation of a circuit relies upon a fixed network of communication channels to route messages from one logical process to another. Other domains are dynamic; the classic "colliding pucks"[46] being an example of this. The traditional approach in such cases is to reduce the communications graph to one which is static and to work from there. In the case of the colliding pucks, the space in which the pucks move is divided into fixed regions (with fixed boundaries) and the simulation is based on that static grid of spaces. The abstraction away from the objects involved in the simulation and the imposition of a more abstract object (the grid of cells) could be avoided with dynamic communications.

Local control: In any distributed system the use of a central control will, ultimately, become a bottleneck in the system. This is true even if the global control is distributed because, as the system grows in size, it will take increasing amounts of time to perform the global operation.

Memory efficient: Memory is relatively cheap and modern machines come with many times the memory available ten years ago. However, ultimately the amount of memory is still a limited resource which needs to be husbanded and allocated sparingly. If a system is not efficient in its memory use, there is also the possibility that accessing the data in memory will take up an increasing amount of time, either because the data is held in virtual memory which needs to be paged in from disk or because the data structure holding the desired information takes times to traverse to locate the actual piece of data desired.

Processor efficient: Processor power is increasing but so is the expectation of what that resource can do for the user. It is important therefore, that the simulation system is efficient in the use of what processor power is available.

Deadlock free: By this we mean that the simulation system should not introduce deadlock where none exists in the real system. Should the protocol under which the simulation is being performed be susceptible to deadlock then steps must be taken to either prevent or to detect and resolve deadlock within the protocol. Any such activity will introduce an overhead into the system which detracts from the system performing useful simulation work.

1.7 Problem to be addressed in this thesis

In this thesis we shall develop a simulation system which has the following features:

- Dynamic communications
- Local control
- Resource efficient
- Deadlock free
- Conservative

- Distributed
- Able to exploit both temporal and structural parallelism

We shall show models of the upper bound of resource consumption (both processor time and bandwidth) as well as experimental results for the simulation of a number of synthetic and real-world systems.

In this chapter we have surveyed the state of the art in distributed simulation and covered the characteristics of the two main systems (conservative – Chandy-Misra-Bryant and optimistic - Time Warp). We note that neither of these systems has all of the attributes of the desired simulation system.

In the next chapter we outline a system which has the desirable features listed above and set the context in which the system was developed.

Chapter 2

Background

In Chapter 1 we looked at the advances made in parallel and distributed simulation from the early conservative CMB systems through the optimistic, or Time-Warp, systems to the various attempts to unify the simulation synchronisation process. We ended the chapter by outlining, and motivating, the features desired in a distributed simulation system.

In this chapter we step back from simulation and look at the more generic problems of the production and synchronisation of data in distributed systems, and how it relates to the desirable features listed in Table 1.3.

2.1 An Approach to the Obtaining the Desirable Features

In conservative CMB-style simulation each LP in the system *must* determine its state for every moment in the simulation. The optimistic Time Warp approach requires that every LP determine its state for every moment during the simulation but also permits the LP to process messages out-of-order and thus potentially erroneously. Should a causality error occur, the LP is then forced to re-evaluate some of its history. Thus, a LP can do more computation than is necessary. This style of processing has been defended by saying *"whenever rollback occurs, other rollback-free implementations would require blocking for an amount of real time equal to that spent on wasted computation"*[45]. In other words, no "useful" time was wasted. While this held true when the user was allocated a specific number of processors for their sole use, increasingly, parallel machines are being created

which permit the sharing of processors by more than one user. The obvious example of such a system is a network of workstations. In these shared multiprocessor systems, computation time taken by one user is denied to another.

Furthermore, it is not always necessary for every LP to know the state of all its inputs for the output result to be determined. Consider, for example, the multiplication of two numbers. Should one number be known to be zero then the other number need not be determined as the result is also zero. Any processor time not used by one user is available to another. The trade-off at issue is the speed of completing one job versus the total throughput of the system.

The problem therefore is how to design a simulation system which permits the result to be obtained by computing only the necessary values.

2.2 Distributing data

The core feature underlying all of distributed computation and, specifically, distributed simulation is the ability to coordinate the production, delivery, and consumption of data[1]. We will use these aspects to derive a protocol with some of the desired features listed in Table 1.3. By "data" we mean discrete packets of information which are complete within themselves.

The problem of distributing data can be split into two separate parts: where to send the data (distribution) and when to create and send the data (production). The case where the location and the time are not independent can be addressed by sending the data to a redirection process at a fixed location which then forwards the data to the appropriate location. This thus reduces the problem to the first case. Maintaining the redirection, or directory, service is outwith the scope of this thesis and does not form part of the argument.

[1]There are some systems in which the data remains in a fixed location and the processing code moves to it rather then the other way around. An example of such a system could be an image analysis application in which, due to the amounts of data to be processed, it is easier for the desired transform to be sent from processing element to processing element than for the image data to be sent.

2.2.1 Data distribution

All data which is to be distributed must have some condition attached to indicate when the data has either reached its desired destination, or is to stop looking. What this implies is that the destination of the data packet must be known before it is sent and is thus under the control of the sending process.

In the case of conservative algorithms, the sending process knows the destination as the topology of the processing elements is fixed. Data packets, once produced, can only be sent to a subset of the processing elements.

In the case of the optimistic algorithms, the sending process knows the destination as it knows the location of all the processing elements in the system. Data packets, once produced, can be sent to any of the processing elements.

While the optimistic algorithms provide for a dynamic topology, the conservative systems do not. One solution to the problem of providing a dynamic topology to the conservative algorithms would be to have a completely connected set of processes and to have each process broadcast its data packet to the rest of the system. While such a system might work, it is impractical as the number of connections needed would grow exponentially in the number of processing elements in the system. What is needed is some way of the sending element knowing which of the possible elements in the system need to receive the data.

2.2.2 Data production

The question also arises of when to produce a new data packet for distribution. The conservative system will only create a data packet when it has sufficient information to determine the contents of that data packet. An optimistic system, on the other hand, will only create a new packet if a change to one of the input values results in a change to the output value. For the sake of simplicity, we will ignore the messages created to effect a rollback should a causality violation occur.

Both of these systems generate data for distribution irrespective of whether or not the data is required by the rest of the system. What is needed is some way for an element in the system to indicate from which other elements it needs fresh data.

The adoption of a publisher/subscriber model of communication is one potential solution to this problem. In a publisher/subscriber model, clients address messages to a topic. Publishers and subscribers are generally anonymous and may dynamically publish or subscribe to the content hierarchy. The system takes care of distributing the messages arriving from a topic's multiple publishers to its multiple subscribers. Such a system is shown in Figure 2.1.

Publisher/subscriber messaging has the following characteristics:

- Each message may have multiple consumers.
- Typically, topics retain messages only as long as it takes to distribute them to current subscribers.

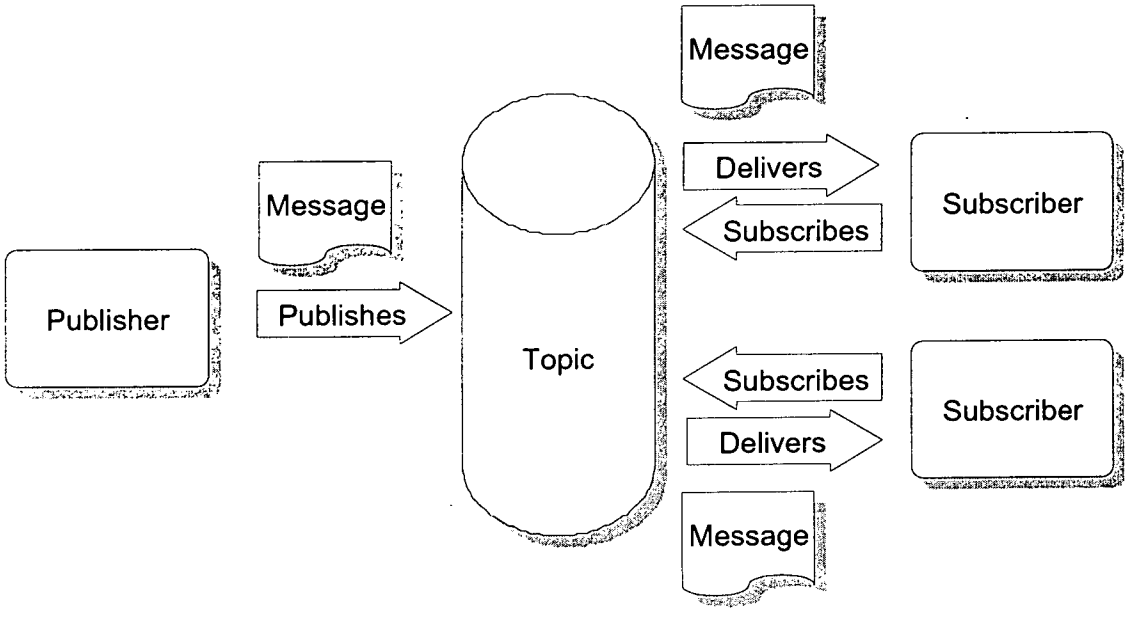


Figure 2.1: Publisher – subscriber communications with a single publisher

Generally there is a timing dependency between publishers and subscribers, because a client that subscribes to a topic can consume only messages published after the client has created a subscription, and the subscriber must continue to be active in order for it to consume messages.

The Java Message Service[62], amongst others, relaxes this timing dependency to some extent by allowing clients to create durable subscriptions. Durable subscriptions can receive messages sent while the subscribers are not active. Durable

subscriptions provide the flexibility and reliability of queues but still allow clients to send messages to many recipients. While most messaging systems, not unsurprisingly, use real time as the temporal measure when assessing whether or not a subscriber can receive a particular message, there would be no great difficulty in using the simulation (or virtual) time in the same way. This would permit a client to subscribe to the topic for a virtual time interval.

2.2.3 A potential solution

From the previous section we saw that a system where the individual elements could indicate from which elements they required data, and also when they required that data, would meet some of the requirements of our "ideal" system; providing a dynamic topology as well as local control. We have, in effect, added the ability to provide a dynamic topology to a conservative system.

Now that a dynamic topology is available, a potential solution presents itself from the wording of the problem. We require a system which would only compute those values which were *necessary* to determine the result. The first step would therefore be to decide what values are necessary and then to determine those values. This leads to a reversal of the standard data-driven method whereby the data is produced and promulgated with the assumption that it will be necessary. Such systems, known as demand-driven systems, have other properties which will be expanded on in Chapter 3.

2.3 Related work

There is little related work addressing simulation *per se*. Demand driven evaluation, and its close relation lazy evaluation, have been studied at various levels from instruction level, through compiler level, to user exposure in languages. In this section we will look at each of these levels.

2.3.1 Request Driven v's Demand Driven

The terms "request driven" and "demand driven" are sometimes used by different authors to mean the same thing. This is unfortunate as they are also used, by

other authors, to indicate different methods of computation.

In order to avoid adding to this confusion, we will clarify what we mean by request and demand driven evaluation.

Request driven: In a request driven system, the client requests that the data be provided *when it is available.* In effect, it notifies interest in the result when it becomes known. This is the default mode of operation of the publisher/subscriber model.

Demand driven: In a demand driven system, the client demands that the data be provided *as soon as possible.* If the data is available, then this is the same as request driven. Should the data not be available, a demand driven system requires the publisher to take action to produce the data (probably by issuing demands of its own).

The rest of the thesis will focus on demand-driven evaluation and related ideas.

2.3.2 Micro level

The dataflow model was originally proposed in the mid-60's. Initially the concept of dataflow was expressed as a graph, which later became parallel program schemes[77, 1, 25]. It was later at MIT[26] that designs of actual computers based on the dataflow model were attempted. Dataflow programs can be described in terms of directed graphs expressing the flow of data between nodes of the graph, with a node representing an instruction or a group of instructions[24]. Data are active and flow asynchronously through the program. The original dataflow model exploits very fine-grain or instruction level parallelism.

The performance of pure, fine-grained, dataflow systems was not able to compete with von Neumann processors[67] when executing sequential programs. Arvind[4] identified the real benefits of dataflow systems as cheap synchronisation and tolerance of memory latency. Hybrid processors that combine features from both von Neumann and dataflow architectures have been developed[68].

The reduction machine[89] is an architecture closely related to the dataflow model. Reduction is based on the demand driven principle and supports func-

tional languages. Beginning at the outermost expression of a functional program, sub-expressions within an enclosing expression are recursively reduced upon demand for their results, into simpler forms, until the expression cannot be further reduced (known as normal form). The reduction process involves rewriting reducible expressions by others with the same meaning until a constant expression representing the result of the program execution is reached. This contrasts with the data-driven model (upon which dataflow is based) which starts execution on the innermost expressions that have their values and propagate the results into the expressions requiring them. One can view the demand-driven and data-driven program graph for the same expression as being identical, except that the direction of the links is reversed. The execution graph of a demand-driven evaluation is dynamically changing during execution, whereas the program graph of the dataflow evaluation is static.

2.3.3 Compiler level

The use of demand-driven (lazy) evaluation can be at a higher level than the processor. A number of languages have been designed to take advantage of lazy evaluation without providing specific constructs to the user. In general, such languages were categorised as non-strict.

In a non-strict language, the arguments to a function are not evaluated until their values are actually required. For example, evaluating an expression of the form $f(exp)$ may still terminate properly, even if evaluation of exp would not, if the value of the parameter is not used in the body of f. Miranda and Haskell[10] are examples of this approach.

In a strict language, the arguments to a function are always evaluated before it is invoked. As a result, if the evaluation of an expression *exp* does not terminate properly (for example, because it generates a run-time error or enters an infinite loop), then neither will an expression of the form $f(exp)$. ML and Scheme are both examples of this.

There is much debate in the functional programming community about the relative merits of strict and non-strict languages. It is possible, however, to

support a mixture of these two approaches; for example, some versions of the functional language Hope do this.

2.3.4 Language level

Halstead[40, 39] proposed a language construct called a *future*. The construct allows programmers to explicitly expose parallelism, with minimal effort, in applicative languages such as MultiLisp. The form (future X) immediately returns a **future**, and creates a task to evaluate X. Rather than waiting for the result of such a computation, the program receives a "placeholder" for that result and is able to continue executing. The placeholder behaves just like any other variable until an attempt is made to use its value; at that point, if the computation is not finished then the thread of execution trying to obtain the value will be blocked until the value is ready.

The principal design rationale behind futures, stated by Mohr et al.[61], is that "the programmer takes on the burden of identifying *what* can be computed safely in parallel, leaving the decision of exactly *how* the division [of work] will take place to the runtime system". The Mohr paper goes on to discuss *lazy* task creation which would be evaluated when needed, which brings us back round to demand driven systems.

The future construct is no longer restricted to the applicative programming domain. Wagner[92] has created portable futures in C++.

It is interesting to note that the simulation of logic circuits was used as a test example to show the power of this construct on a multi-processor machine[11].

2.3.5 Demand driven Simulation

Most of the papers dealing with demand driven simulation have been quite firmly rooted in the domain of logic simulation. A number of them state that demand-driven evaluation can be easily expanded to a larger class of systems but fail to address the issues involved, such as function re-evaluation and random number generation. The results of most functions are deterministic. Random number servers are, hopefully, non-deterministic. Any attempt to re-evaluate a random

Circuit	"unoptimised" evaluation elapsed time ratio	"optimised" evaluation elapsed time ratio
74181 ALU	0.434	0.238
C432	0.605	0.284
C499	0.647	0.525
C880	0.607	0.463
C1355	0.747	0.616
C1908	0.864	0.637
C2670	0.954	0.534
C3540	1.296	0.398
C5315	1.247	0.444
C7552	1.337	0.740

Table 2.1: Ratios of Demand driven simulation to event driven simulation

number function **must** produce the same number as the initial request.

The earliest paper that can be found relating demand-driven evaluation and simulation is by Smith et al.[81]. The paper presents a sequential algorithm for the simulation of digital logic circuits and compares it with a standard event driven algorithm. The test circuits used are those which were created for the *International Symposium on Circuits and Systems, 1985*[12] which have since become the closest thing to a standard benchmark circuit that the logic simulation community has. Two different evaluation models are presented. The first is a standard event driven evaluation model. The second has a number of optimisations applied; most notably early cutoff (evaluation stops as soon as the result is known). The optimised system consistently out-performs the standard event-driven approach. A table of the results is shown in Table 2.1.

It should be noted that this paper, and other results by the same author, describe a system which uses requests, not as the main driving force behind the simulation, but as a way to ensure that the simulation proceeds; should an input not have a current message, a request is sent asking for the data to be forwarded.

A second report by Smith[80] expands on his earlier work and includes, for the first time, a notion of time windows which encompass a number of discrete-event time units. Using the same test circuits as were used in the earlier paper described above, a number of experiments were conducted to determine the effect of various

modification to the algorithm. As a base case for comparisons, the system is compared with a standard event driven algorithm where both simulators use the same evaluation routine. The demand driven system is then modified to include early cutoff in its evaluation. A further experiment is made using a number of heuristics to reorder the input pin evaluation order.

A further modification of the algorithm is given which provides methods for modelling both transport and inertial delays in logic circuits. This information is presented in a more general form by Charlton[22] which we will cover later.

Subramanian and Zargham[87] move demand-driven simulation explicitly into a parallel arena for the first time. Three different algorithms are used for comparison purposes. The first of these is a discrete event system similar to the CMB algorithm[19]. The second is a pure demand-driven system (DD) and the third is an algorithm with two phases: the earlier phase determines which input values will be needed using a demand-driven approach and the later phase then evaluates in a standard CMB manner. The results show that the pure demand driven system performs better than either the CMB system or the two-phase system. The circuits used are not described or attributed with only the number of elements used and the type of the circuit (sequential or combinatorial) stated.

There has been some work performed by Charlton et al. on demand-driven simulation of logic circuits[21]. Most of their work derives from studies of lazy evaluation. The earliest paper investigates the effect of differing event scheduling strategies for both demand and event driven systems. It shows that significantly fewer events need to be processed with a demand driven system. The results are based on a uniprocessor simulator with a single queue.

Charlton[22] demonstrates a method for modelling general delays in demand-driven simulation. Basically, assuming a node has a maximum (minimum) delay of t_{max} (t_{min}) then a request for data for an interval (a,b) is fulfilled by a request for data in the interval ($a - t_{max}$,$b - t_{min}$) as this interval will always be sufficient to determine the input values for whatever the actual delay turns out to be. This work is presented in relation to a system which is being driven by time-stamped requests and not intervals.

Both of the above papers have addressed issues from the perspective of uniprocessor simulation. Another paper from the same group[30] covers parallel evaluation strategies for demand-driven simulation. Optimal evaluation orderings are obtained for a number of basic two input logic functions assuming that two processors are available to evaluate the function. Heuristics are then developed which make implementation more practical. Results obtained show that the heuristics perform no worse than 1.6 times slower than the optimal strategies.

Only one of the papers surveyed deals with the evaluation of abstract simulation models[63]. The models are built in Miranda and then different simulation evaluation strategies are applied to the models. It concludes that the demand driven system is inefficient in both space and time and that a discrete event (data driven) system can deal with all inefficiencies. It also states that the demand driven system is less expressive as it cannot model inertial delays. This has been shown to be incorrect in the work of Charlton[22]. It closes by stating that as demand driven systems are inefficient they are easy to parallelise. A number of the criticisms targeted at demand driven simulation are addressed in Chapter 3.

2.4 Speedup and Efficiency

When assessing the quality of the performance of a parallel system, two measures have often been used. The first, speedup, indicates how much faster the result is obtained as the number of processes increase (Equation 2.1). The numerator, T_1, is sometimes the time taken by the best possible sequential solution, but is generally taken to be the time for the parallel code running on a single processor. The denominator, T_n, is the time taken when using n processors.

$$S(n) = \frac{T_1}{T_n} \tag{2.1}$$

The second measure, efficiency, is defined as the average utilisation of the n allocated processors. Ignoring I/O, the efficiency of a single processor system is equal to one. The relationship between speedup and efficiency is given by

$$E(n) = \frac{S(n)}{n} \tag{2.2}$$

Eager et al.[32] argue that these measures can be used to determine an "optimal" number of processors to be used in the execution of a given problem. They plot a measure of "benefit" (execution time) against a measure of "cost" (number of processors) and note that a *knee* occurs in the graph. The knee is the point where the benefit per unit cost is maximised and which, intuitively, represents an optimal system operating point. They argue that being able to estimate the number of processors that yields the knee is important as that would indicate the appropriate allocation of processors for that job.

While these measures have relevance in the multi-programmed, multiprocessor with a static processor allocation, their use in a network-of-workstations environment is less clear as they only take into account real cost (the resources consumed).

2.4.1 Opportunity cost

Opportunity cost is a basic term from the disciplines of economics and accounting. In these circles the acceptable definition of the term is, "the advantage forgone as the result of the acceptance of an alternative".

In assessing the efficiency of a system the opportunity cost has often been ignored. This stems from the understanding that a system needs certain resources before it can start and holds them until it is finished. In machines with a static allocation of processors, this is the natural state of affairs. Such a situation does not hold true in multiprocess systems where an individual process will only reserve a resource for the duration needed. It might claim and release that resource many times during the its lifetime. An example would be time-slicing CPU access on a multiuser system.

Opportunity cost is a relative measure in that it compares what is with what could have been. As such care must be taken in its use to avoid trying to compare the incomparable.

2.5 Binary Decision Diagrams

Before we leave this chapter, it is useful to take a look at how boolean functions can be represented using binary decision diagrams (BDD)[14], and in particular, Ordered Binary Decision Diagrams (OBDD)[15]. It should be noted that the use of BDD is not needed for a demand-driven system to function, but they can be used to make clear some of the benefits of a demand-driven system.

Binary decision diagrams have been recognised as abstract representations of Boolean functions for many years[3]. A binary decision diagram represents a Boolean function as a rooted, directed acyclic graph.

A binary decision *tree* is formed by expanding the binary expression around a single variable and then repeating for each sub-expression until there is no expression left to evaluate.

Consider the expression

$$f(A,B,C) = \overline{A}B\overline{C} \vee AC$$

This expression could be expanded as follows[2]:

$$f(A,B,C) = A(f(1,B,C)) \vee \overline{A}(f(0,B,C))$$

This could be repeated for the variables B and C.

A graph of the resulting tree is shown in Figure 2.2.

Each non-terminal vertex is labelled with a variable $var(v)$ and has arcs directed to two children: $lo(v)$ corresponding to the case where the variable is assigned the value 0 and $hi(v)$ corresponding to the case where the variable is assigned the value 1.

2.5.1 Reducing the tree

This naïve representation provides 2^n paths from the root to the leaves, one for each of the 2^n different combinations of input values. A number of reductions can be applied to reduce the number of available paths.

[2]This identity is known as the *Shannon expansion of f* with respect to A, although it was originally recognized by Boole

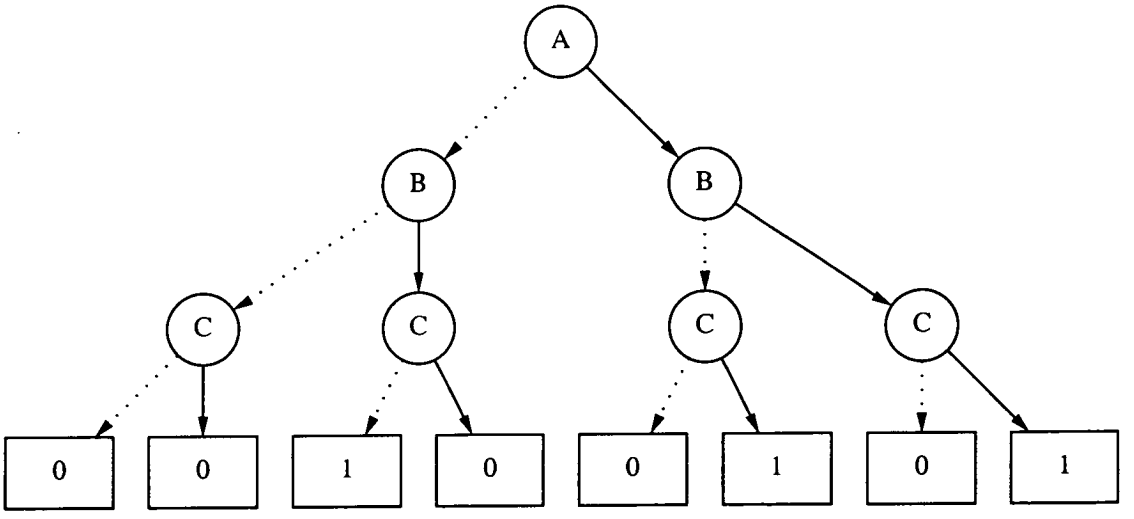


Figure 2.2: Decision tree for $f(A, B, C) = \overline{A}B\overline{C} \vee AC$

There are three transformation rules which can be applied to the graph without altering the function being represented:

Remove duplicate terminals. Eliminate all but one vertex with a given label and redirect all arcs into the eliminated vertices to the remaining one.

Remove duplicate non-terminals. If the non-terminal vertices u and v have $var(u) = var(v), lo(u) = lo(v)$, and $hi(u) = hi(v)$, then eliminate one of the two vertices and redirect all incoming arcs to the other vertex.

Remove redundant tests. If non-terminal vertex u has $lo(u) = hi(u)$, then eliminate vertex u and redirect all incoming arcs to $lo(u)$.

Considering again the graph (Figure 2.2) we can see that the leftmost C node has a redundant test and can thus be removed and replaced with the constant 0. Similarly, the two rightmost C nodes are identical and we can thus remove them using the **remove duplicate non-terminals** rule. Lastly, we can see that the rightmost B node can now be removed using the **remove redundant non-terminal** rule. This leaves the tree as shown in Figure 2.3. By applying the **remove duplicate terminals** rule we get the decision diagram as shown in Figure 2.4.

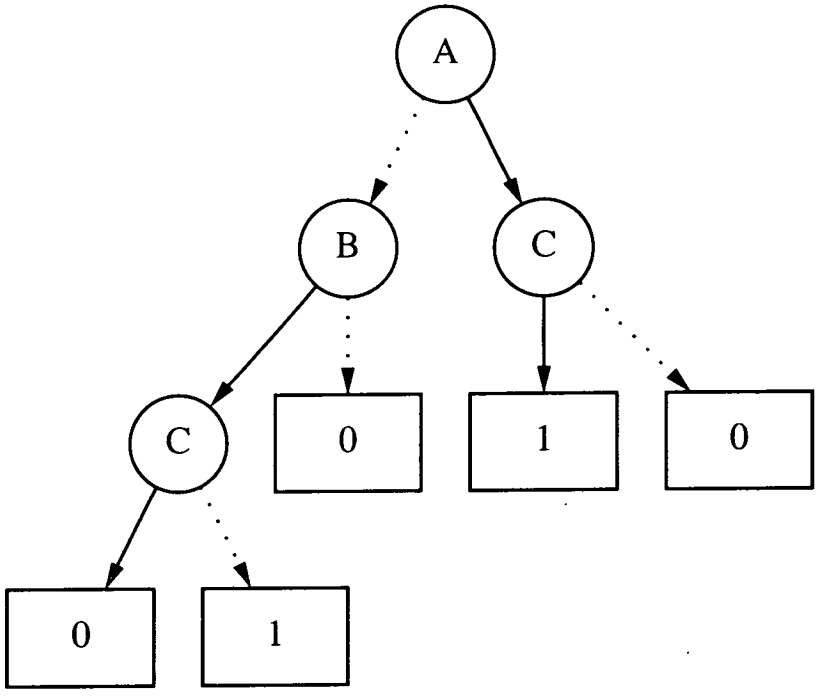


Figure 2.3: Fully reduced decision tree for $f(A, B, C) = \overline{A}B\overline{C} \vee AC$

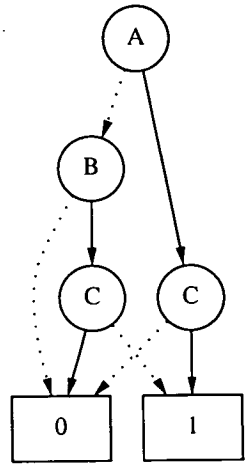


Figure 2.4: Decision diagram for $f(A, B, C) = \overline{A}B\overline{C} \vee AC$

2.5.2 Combining diagrams

Each binary decision diagram represents a boolean function and can be created by combining simpler boolean functions in the manner described below.

First, an explanation of some notation. Consider a function f which takes a vector $\vec{x}$ then the notation

$$f|_{x_i \leftarrow 0}$$

means the function with the value of x_i set to the constant 0. This is sometimes referred to as *restriction.*

Now, the combination of two functions f and g by the operation $< op >$ can be defined as

$$f < op > g = \overline{x}.(f|_{x \leftarrow 0} < op > g|_{x \leftarrow 0}) + x.(f|_{x \leftarrow 1} < op > g|_{x \leftarrow 1})$$

This technique will provide an algorithm for computing $f < op > g$ with a time complexity which is exponential in n (the number of inputs). There are various methods and improvements to the algorithm which reduce this complexity[14, 15].

2.6 Attributes of Decision Diagrams

Decision diagrams have a number of attributes which make them useful in demand driven evaluation.

2.6.1 Automatic short circuiting

Functions can be split into three classes.

Strict: These functions **always** require all of their inputs before they can evaluate an output, e.g. addition.

Partially strict: These functions **may** require all of their inputs before they can evaluate an output, e.g. multiplication (when one input is zero).

Non-strict: These functions **never** require all of their inputs before evaluating an output, e.g. if...then...else... Either the *then* branch needs to be evaluated **or** the *else* branch, but never both branches.

Short circuit evaluation is equally applicable to both data and demand driven systems. It allows the result of the function to become available as soon as possible. Consider the binary decision diagram for a three input AND gate (Figure 2.5). If any of the inputs evaluates to 0 then the result is known. In a data driven system this would mean that the result can be available before all the inputs have evaluated. In the demand driven system it can mean a reduction in both communication and computation as we will see later.

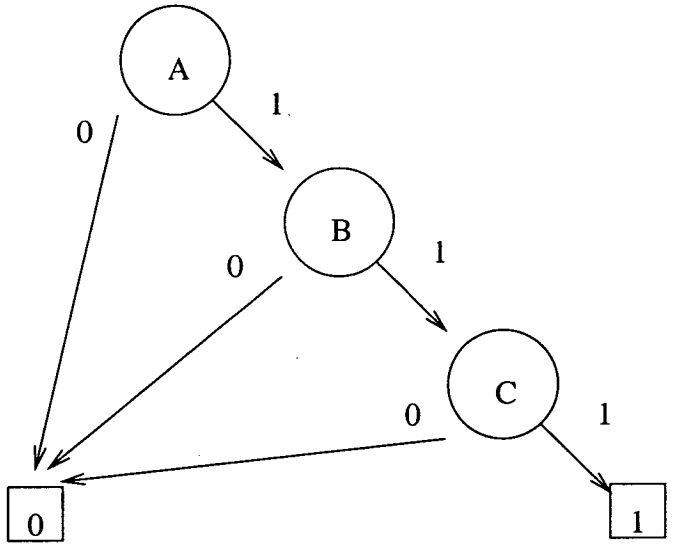


Figure 2.5: A decision diagram for a three input AND gate.

The early interest in demand driven systems for logic circuits might be explained by the fact that of the 16 two-input boolean functions only 2 are strict and the rest are partially-strict.

An interesting aside which arises from the use of binary decision diagrams is the fact that the resultant diagram implements 'short circuit' evaluation. As such, the data which is required to evaluate the function depends on the values of that data which has already been requested.

Further, it is possible to generate the remaining function by restricting the current function by any of its variables in any order. For example, consider the function used earlier ($f(A, B, C) = \overline{A}B\overline{C} \vee AC$).

$$f|_{C \leftarrow 1} = A$$

2.6.2 Maximal request set

Automatic short-circuiting of BDD means that one can find the maximal set of variables which **must** be determined to evaluate the function. This can be found by first determining the nodes which require to be evaluated down each path in the diagram. The maximal set is the intersection of all these path sets and will always contain the root node at least. Once a value has been returned from the initial set of requests, it is possible to recompute the maximal set and issue requests for the value of any node which was not in the original set but which is now included.

2.6.3 Reduction in false negatives

In the section above we have concentrated on binary decision diagrams. There is, however, nothing in the formulation of the equations or systems which prevents ternary functions being defined and manipulated. So, in the case of logic gates, a three-valued system is often used with one value being X or unknown[3]. In such systems each node has three output arcs instead of the conventional two. Full details of these systems can be found in [49].

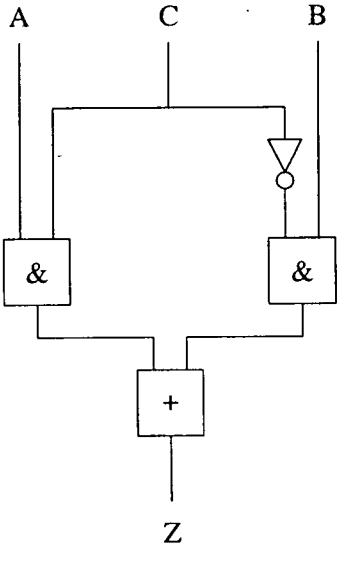


Figure 2.6: Gate level implementation of a 2-1 multiplexor

[3]Note that this value is not an intermediate value somewhere between low and high. Rather it is an indication that the variable has the value of either low or high but we cannot determine which.

Consider a simple 2-1 multiplexor created using this simple logic system. The output Z can be given by the following equation (being derived directly from the physical implementation shown in Figure 2.6).

$$Z = AC \vee B\overline{C}$$

If the value of C (the control variable) is set to unknown, then the above equation will indicate that the output is also unknown. This is, at first glance, a reasonable result since we cannot determine which of the two inputs should be allowed to continue to the output. However, on reflection it is less reasonable; if both inputs have the same value then irrespective of *which* is allowed to continue the output would have the same value as either of the inputs. Should this simple multiplexor be simulated as a gate level implementation then it could inject false-unknowns into the network.

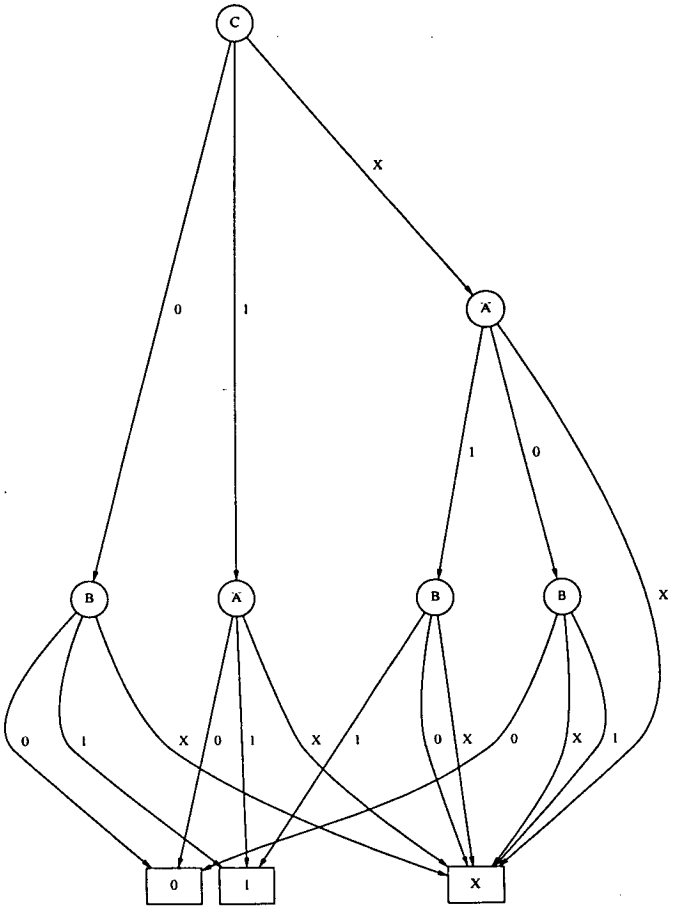


Figure 2.7: Decision diagram for 2-1 multiplexor using a three valued logic

However, should the multiplexor be implemented as a single unit with a function derived from the decision diagrams for the three gate implementation, these false-unknowns will not occur. The diagram is shown in Figure 2.7.

2.7 Chapter Summary

In this chapter we stepped back from simulation and looked at the more generic problems of the production and synchronisation of data in distributed systems. We saw that, by making the receiver responsible for requesting the data rather than have it wait passively, we could obtain a dynamic topology for inter-element connections.

We clarified the difference between request driven and demand driven system and looked at related work at the micro, compiler and language levels. Work in simulation, using either demand driven or request driven systems, was also discussed.

We looked at the definitions of speedup and efficiency that are widely used throughout the distributed systems field as quality measures and proposed a new measure, opportunity cost, which gives an indication of how much of the systems resources are withheld from other potential users.

We closed the chapter by looking at binary decision diagrams. This method of expressing a function can make explicit any non-strictness in that function and as such, is well placed to be utilised in a demand driven system.

In the next chapter we take the results from our investigation and propose a demand driven simulation system.

The insights obtained will then be used to address the requirements of our desired simulation system.

Chapter 3

Demand-Driven Simulation

This chapter discusses some of the costs and benefits associated with demand-driven simulation. The costs are resource consumption, be they bandwidth, processor or time. It provides arguments in mitigation of a number of the costs involved as well as strategies to reduce the overall cost of simulating a system.

3.1 Costs and Benefits of Demand-Driven Simulation

Parallel discrete event simulation has been data driven since its inception. This can be considered a logical progression from the sequential simulation systems where there existed a queue of events which needed to be processed. The key word in the previous sentence was "needed". One problem with data driven systems is that events will be generated which will have no effect on the receiving node and therefore, for efficiency reasons, need not have been generated in the first place. Unfortunately the logical process which generated the event could not have foreseen this and hence the event has to be generated "just in case".

The underlying idea is that the system under simulation is a "black box" which is exercised by a series of data values and the changes in outputs are observed. Demand-driven systems reverse this concept; the output is interrogated and demands for data propagate up the system and back in simulation time until a demand is made of the input data pool. Since in a demand driven system logical processes only generate an event when the receiving node requests it, potentially only the minimum amount of work needs to be performed.

The biggest potential problem with standard demand driven systems is that for each event needed *two* messages must be sent; one to request the event and the event itself. In parallel systems, where communications are generally much slower than computations, this could appear to be a significant problem. The effect might be mitigated by the decreased computation required as only necessary work is performed. We term this "the doctrine of necessary computation".

3.1.1 The Costs

A number of costs are associated with demand driven simulation which are not associated with data driven simulation. In this section these costs are analysed and evidence presented of ways in which these costs might be mitigated.

3.1.1.1 Communication costs

There is an obvious problem with such a **demand driven** system: extra communication. The most glaring inefficiency, and the one which is most often stated as a reason for not using this method, is the extra communication which is required to "spark" the computation. In the worst case each data value will require twice the number of communications than it would under a data driven system. A point to make about this observation is that it is the *number* of communications which would, at worst, double and not the communication load itself. If we use the simple equation $\alpha + \beta l$, where α is the start-up cost for communication, β the cost per unit transmitted and l is the message length then the actual overhead is:

$$\frac{2\alpha + \beta(l_d + l_r)}{\alpha + \beta l_d}$$

where l_d is the length of the data message and l_r is the length of the request message.

If we assume that the size of the request message is smaller than the size of the data message then the overhead must be strictly less than a factor of 2. If we assume that a request message is significantly smaller than a data message, the equation can be written as

$$2 - \frac{\beta l_d}{\alpha + \beta l_d} \tag{3.1}$$

This approximation to the overhead factor can be analysed by cases.

$\alpha >> \beta l_d$ The start-up cost is significantly larger than the transport cost. This situation could occur if a large number of small messages were to be sent. The overhead tends towards a factor of 2.

$\alpha = \beta l_d$ The start-up costs and the transport costs are equal. The overhead is a factor of $1\frac{1}{2}$.

$\alpha << \beta l_d$ The start-up cost is significantly smaller than the transport costs. This situation could occur if large messages were to be sent. The overhead tends towards a factor of 1. In other words, as the size of the data message increases, the request induced communication cost as a fraction of the total communication cost falls.

There are further techniques which can be employed to reduce the communication traffic. The simplest technique is to bundle a number of requests into a single message. The practicality of this method will depend upon the time advance mechanism in the simulation. Such a system has been used in data driven time warp simulation by Butler and Wallentine[16]. They show that bundling events into a single message can reduce the communications load but that the benefit varied with cancellation strategy. If we assume that the simulation is interval based, as for ELSA [5, 6], then a number of requests for consecutive intervals can be compressed into a single message no larger than a single request. ELSA is explained in greater detail in Chapter 4.

3.1.1.2 Computation costs

The next cost to be considered is an increase in work required at a logical process. Every message which arrives at a logical process must be handled. This is produced by the requirement to handle incoming request messages as well as data messages. Again, the doctrine of necessary computation can help to reduce this overhead as fewer logical processes need to be evaluated. The result could be that the computation load is concentrated on fewer logical processes than in the equivalent data driven system.

3.1.1.3 Memory costs

Memory requirements may rise because of the requirement of some types of node to store their previously computed data. In the case of a random number generator it might be desirable for every node which requests its state at time t to be given the same answer. The use of MEMO functions has been studied in the realm of functional programming[58, 43]. The idea behind them is very simple: a memo-function is like an ordinary function, but it remembers all the arguments it is applied to, together with the results computed from them. If it is ever re-applied to an argument the memo-function does not recompute the result, it just re-uses the result computed earlier. "Memoisation" is an optimisation which replaces a potentially expensive computation by a simple table look-up.

One of the difficulties in implementing general memo functions arises from the need to determine whether two calls to the same function are equivalent. In a general solution, comparing data-structures for equality is expensive. It is not uncommon for a conservative definition of equality to be used for the test. Two objects could be tested for identity as follows:

1. If they are stored at the same address, then they are identical. Return true.
2. If they are atomic values (such as numbers, booleans, characters) then they are identical if they are equal.
3. Otherwise they are not identical. Return false.

Fortunately in demand driven simulation the inputs to any logical process are tagged with their time. Therefore the task of determining whether two inputs are identical reduces to comparing their associated times. As demand driven simulation has no concept of "fossil collection" which is common in timewarp systems to manage memory, it would be possible for these memo stores to grow throughout the simulation run. This might be desirable as it would provide a record of the state of the logical process throughout the simulation. This may be of great use in a postmortem of the simulation run.

However, in cases where we are only interested in the final value and not in the intermediate results we can consider these memo stores to be, in effect,

caches. As such certain cache replacement algorithms could be employed to limit the memory requirement. The commonly applied LRU algorithm where the least recently used block is replaced might be used effectively, but this requires further investigation.

3.1.2 The Benefits

The use of demand driven simulation is not without qualitative benefits. An important benefit is that the logical processes in the system under consideration are able to dynamically reconfigure their local network in order to contact any other logical process. Below we describe five benefits of using demand-driven evaluation.

"Necessary" Computation: each node in the system makes only those requests which it deems necessary. This means that if the value of an input is not required then it is not evaluated. What input values are required can depend on the values of the other inputs. For example, (A or B), if A is evaluated and returns 0 then B must be evaluated. If B returns 1 then the answer could have been found by only evaluating B. So "necessary" computation does not mean minimal computation.

Dynamic Interconnect: the requesting logical process (LP) is free to communicate with any other LP in the system. As such the connections are generated at run time and are able to respond to data dependent conditions. This could be considered as a global all-to-all topology but with the addition that each LP is only dependent on a certain sub-set of processes and are not constrained to the "lock step" that such a configuration would tend to produce in data driven systems.

Sub-system Activation: to investigate a part of a system it is now only necessary to send a request to that sub-system. The doctrine of necessary computation results in only those parts of the system which need to be activated being computed.

Realistic Data: one of the major problems in simulating a *component* of a system is generating realistic input data. The difficulty is that realistic data comes from activating the whole system at a specific level of abstraction and this can be too computationally intensive to achieve. In a request driven system only those elements which are required are activated and thus potentially fewer nodes need to be computed to provide realistic data.

Static graph emulation: a graph which is known to be static throughout the simulation run can be modelled. The number of extra messages is proportional to the number of edges in the static system. Each edge is initialised to carry a message requesting data from the start of time to the end of simulation time. The request driven system will now behave as a data-driven system.

The most striking, and potentially the most beneficial, aspect of demand driven systems is that, as data is only produced on demand, only those units whose results are needed by the computation are actually evaluated. The result of this is that the communication and computation costs will be reduced. This will have a knock-on effect on multiuser systems as they will have more resource available to them for their tasks. The more unnecessary work that can be avoided the higher the opportunity cost saving compared to data driven evaluation of the same system.

3.2 Strictness and Threshold Functions

If all the functions being evaluated by logical processes were strict – each always needing all their parameters – then the scope for opportunity cost savings would be reduced (or eliminated altogether). In order to control the strictness more easily and to evaluate the system for generic functions, it was decided to use threshold functions.

3.2.1 Threshold Functions

Threshold functions appear in a number of different fields[64] and can exist in a number of different forms[59]. The output of a threshold function is some function of the sum of its weighted input values. Consider a function with n inputs. The intermediate value is given by:

$$Y = \sum_{i=1}^{k} w_i \times X_i$$

where X_i is the value of input i, and w_i is its weight. The final value is given by some function $F(Y)$. Different applications of threshold functions use different decision functions F. We will be using unit weighted, hard threshold functions.

$$F(Y) = \begin{cases} 0 & \text{if } Y < k \\ 1 & \text{otherwise} \end{cases}$$

The threshold value is k. We denote a threshold function of n inputs and a threshold of k by T_k^n.

3.2.2 Strictness

We define the strictness of a function as the amount of data which the function requires, on average, to compute the result. The value of strictness lies in the interval $[0 \ldots 1]$. A strictness of 0 would indicate that the function requires no external information to compute a result. This is equivalent to a constant function. A strictness of 1 would indicate that the function requires all of its inputs to be evaluated, every time, before it can correctly generate an output.

The strictness of a function, f, with n inputs, is defined below.

$$M(f) = \frac{C(f)}{n}$$

The function $C(f)$ is the average number of inputs which are evaluated to compute the function f. If the value of $C(f)$ depends on the order in which the inputs are evaluated then the strictness measure calculated is with respect to a particular variable ordering. An overall strictness can be calculated by a weighted average of this value over all possible orderings of the input variables.

3.2.3 Determining C for Threshold Functions

We shall assume that each input is equally likely to evaluate to 1 as it is to evaluate to 0. Further, we shall note that the order in which inputs are evaluated does not effect the count calculated.

There are four cases to consider.

1. T_k^0 – No further inputs need to be computed and therefore the result must be known. $C(T_k^0) = 0$.

2. T_0^n – As the function can be computed when k inputs have evaluated to 1 and the value of k is zero, then the function need not evaluate any more inputs. $C(T_0^n) = 0$.

3. $T_k^n, n < k$ – As there are no longer sufficient inputs remaining to be evaluated to possibly satisfy k then the function must evaluate to 0 without further work. $C(T_k^n) = 0, n < k$.

4. $T_k^n, n \geq k$ – There are n inputs remaining to be evaluated and once an arbitrary input has been computed then the remaining patterns will be split into two sections. The cost of evaluating those sections must also be included. $C(T_k^n), n \geq k = 1 + \frac{C(T_{k-1}^{n-1}) + C(T_k^{n-1})}{2}$.

$$C(T_k^n) = \begin{cases} 0 & \text{if n=0 or k=0 or } n < k \\ 1 + \frac{C(T_{k-1}^{n-1}) + C(T_k^{n-1})}{2} & \text{otherwise} \end{cases}$$

Should an input favour either a zero or a one then the function for $C(T_k^n)$ needs to be expressed in a more general form.

$$C(T_k^n) = 1 + P(I_i = 1)C(T_{k-1}^{n-1}) + P(I_i = 0)C(T_k^{n-1})$$

where $P(I_i = 1)$ is the probability that input i evaluates to 1.

Implicit in the above formula is an ordering on the evaluation of the inputs. When the probability of a one or a zero is the same for all inputs then the order is unimportant as the end result will be the same. When the probabilities differ then the result depends upon the order in which the inputs are evaluated. This is

shown by evaluating a T_2^2 function. Let $F(a,b) = C(T_2^2(a,b))$ and $P(a=1) = 0.9$ and $P(b=1) = 0.5$.

$$\begin{aligned} F(a,b) &= 1 + 0.1F(0,b) + 0.9F(1,b) \\ F(0,b) &= 0 \\ F(1,b) &= 1 \\ \Rightarrow F(a,b) &= 1.9 \end{aligned}$$

$$\begin{aligned} F(b,a) &= 1 + 0.5F(0,a) + 0.5F(1,a) \\ F(0,a) &= 0 \\ F(1,a) &= 1 \\ \Rightarrow F(b,a) &= 1.5 \end{aligned}$$

3.3 Input Selection

As we saw above the strictness of a threshold function can depend on the order in which the inputs are evaluated. Demand-driven simulation can benefit from this characteristic through short-circuiting. For example, if any input to an AND-gate is found to be logic 0 then the output from that gate is also logic 0. This is known without needing any other information. Other logic functions have similar properties. If the logic function is expressed as a binary decision diagram, this "short-circuiting" is automatic. Advantage can be taken of these properties by applying a *lazy evaluation* rule. Application of this rule throughout the system will reduce the amount of computation required to determine its output.

The number of activations required to evaluate a function will therefore depend on the *order* in which the inputs are evaluated. This section is concerned with obtaining an ordering of the inputs to a component so as to minimise the expected evaluation time. Two factors may influence the evaluation policy:

1. The likelihood of each signal having a desired value; assuming that this is known *a priori*, or can be calculated (p).

n	k	$M(T_k^n)$
10	5	82.930
10	4	73.906
10	3	58.438
10	2	39.746
10	1	19.980
9	5	83.767
9	4	78.299
9	3	63.715
9	2	43.924
9	1	22.179
8	4	81.738
8	3	69.434
8	2	48.926
8	1	24.902
7	4	83.036
7	3	75.223
7	2	54.911
7	1	28.348
6	3	80.208
6	2	61.979
6	1	32.813
5	3	82.500
5	2	70.000
5	1	38.750
4	2	78.125
4	1	46.875
3	2	83.333
3	1	58.333
2	1	75.000
1	1	100.000

Table 3.1: Percentage Strictness for some threshold functions.

2. The expected evaluation time for each signal input (w).

The ordering problem has been analysed for the case of AND gates[79]. The analysis for OR gates follows easily from this work. The characteristic of an AND gate which we seek to exploit applies, more generally, to any gate with n inputs which requires k or more of its inputs to be logic 1 before its output becomes logic 1, $1 \leq k \leq n$. Such gates are known as *threshold* gates. The corresponding boolean function of n inputs, $X_n =< x_1, x_2, \ldots, x_n >$, is denoted by $T_k^n(X_n)$. Thus T_1^n is the n-input OR function and T_n^n is the n-input AND function. A circuit whose components are all threshold gates (or their negations) is called a threshold circuit.

Dunne and Leng[31] expanded on the work of Sassa and Nakata[79] to provide a general evaluation strategy for threshold circuits. Their work is outlined below.

In order to verify that T_k^n evaluates to a logic 1, at least k inputs must be calculated. The minimum cost of evaluation is obtained by choosing the first k elements from the order:

$$\frac{w_i}{p_i} < \frac{w_j}{p_j} < \cdots < \frac{w_k}{p_k}$$

Equally, in order to verify that the function evaluates to a logic 0, at least $(n-k)+1$ inputs must be calculated. The minimum cost of evaluation is obtained by choosing the first $(n-k)+1$ elements from the order:

$$\frac{w_i}{1-p_i} < \frac{w_j}{1-p_j} < \cdots < \frac{w_k}{1-p_k}$$

Note that there is always at least one input which is a member of both sets. This follows as k elements are in the first selection and $(n-k)+1$ elements are in the second and thus $n+1$ elements are represented in total. As there are only n discrete elements available, at least 1 element must be represented twice. There may, of course, be more than one element represented twice. We shall call the set of such elements the **Common** set.

Once an element from the Common set has been evaluated we are left with one of two threshold functions depending on the value of the calculated input: either, T_k^{n-1} if the value is 0 or, T_{k-1}^{n-1} if the value is 1. The next input to be

evaluated can be chosen in a similar manner until the final output of the function has been determined. In their paper, Dunne and Leng[31] do not give a rule on which element in the Common set to select. Should there only be a single element, then the choice is obvious. However, should there be 2 or more, then it is unclear which will provide the minimum expected cost for evaluating the function. The algorithm permits us to choose any element in the Common set. We see this in the common set of lists 3.2 and 3.3.

If we assume that the probabilities and costs are fixed then an evaluation policy can be determined in advance for each gate in the circuit.

3.3.1 Example

Consider the threshold function T_3^5 with the inputs X_n having the weights and probabilities shown below.

	X_1	X_2	X_3	X_4	X_5
w	100	200	400	200	150
p	0.8	0.2	0.5	0.9	0.3

When sorted the *1-list* and *0-list* appear as shown below. The elements in $\{\ldots\}$ are those eligible to be chosen to be evaluated.

$$1-list \quad \{x_1, x_4, x_5\}, x_3, x_2 \tag{3.2}$$

$$0-list \quad \{x_5, x_2, x_1\}, x_3, x_4 \tag{3.3}$$

Either x_1 or x_5 could be chosen. In this example we shall choose x_1. If x_1 evaluated to 0 the lists would become

$$1-list \quad \{x_4, x_5, x_3\}, x_2 \tag{3.4}$$

$$0-list \quad \{x_5, x_2\}, x_3, x_4 \tag{3.5}$$

If x_1 evaluated to 1 the lists would become

$$1-list \quad \{x_4, x_5\}, x_3, x_2 \tag{3.6}$$

$$0-list \quad \{x_5, x_2, x_3\}, x_4 \tag{3.7}$$

If this is repeated, then the resultant tree is shown in Figure 3.1.

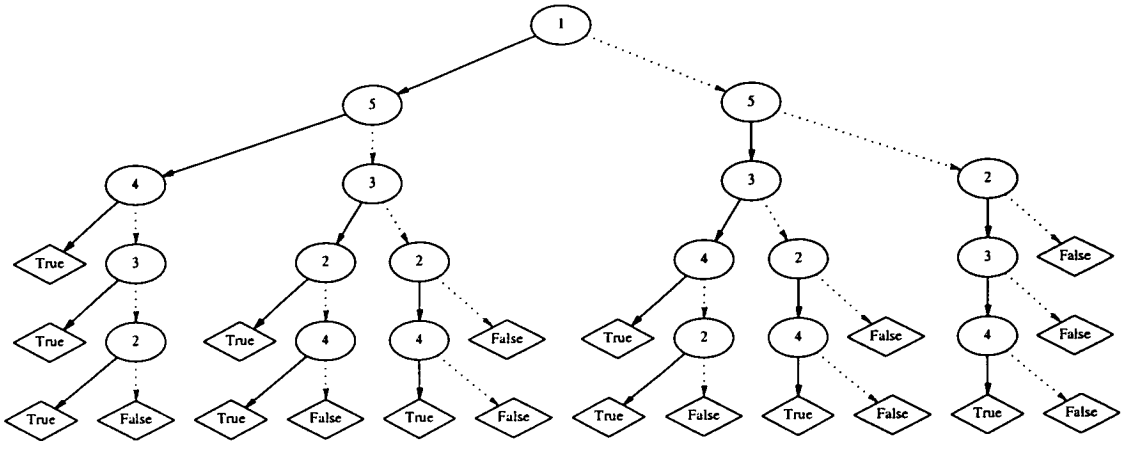


Figure 3.1: A minimum expected cost evaluation graph by the method of Dunne and Leng

3.3.2 Remarks

The method described purports to provide an optimal evaluation policy for simulating gates in threshold circuits under certain assumptions. However, the method is not guaranteed to provide a unique solution and can generate evaluation strategies that, while of low cost, are still sub-optimal. Multiple solutions are available as a result of non-singleton Common sets.

Another major assumption in the method is that the probabilities associated with each input are independent. This is only true in tree shaped circuits. Further it is assumed that the cost of evaluating any input is independent of when it is evaluated. While this is true in "classic" demand driven systems, any system which uses memo[43] functions breaks this constraint.

Looking again at Figure 3.1 we note that the immediate children of the root node are the same. This means that irrespective of the value of input 1 (the root node), the next input in order to be evaluated is input 5. By waiting for the value of input 1 before demanding the value of input 5 is, in effect, serialising a possible parallel operation. We shall look at the opportunities for parallelising the evaluation of a threshold function in Section 3.4.

3.3.3 The enumeration of all possible labelings of threshold trees

The number of valid decision trees for a threshold circuit of size n with threshold k is given by the function below.

```
fun count 0 _ = 1
  | count _ 0 = 1
  | count n k = if (n<k)
                then 1
                else (n*(count (n-1) k)*(count (n-1) (k-1)));
```

For a T_3^5 circuit the number of valid trees is 414720. After evaluation the expected computation time of each of these trees with the parameters in Table 3.2 (the values are from Dunne's paper[31]) we find the frequencies listed in Table 3.3.

	1	2	3	4	5
w	10	100	15	80	63
p	0.04	0.1	0.5	0.8	0.3
$\frac{w}{p}$	250	1000	30	100	210
$\frac{w}{1-p}$	10.41	111.11	30	400	90

Table 3.2: Parameters for a (5,3) Threshold circuit from Dunne's paper.

Interval	Frequency
150-159	1036
160-169	6881
170-179	21618
180-189	21617
190-199	54475
200-209	48276
210-219	48204
220-229	66971
230-239	56358
240-249	57171
250-259	30147
260-269	1966

Table 3.3: The frequency count of expected evaluation costs

All the trees generated by the algorithm, in our experiments, lie in the 150-159 cost band. The minimum graph (found by exhaustive search) is shown in Figure 3.2. It has a cost of 157.032. It is worth noting that this is not the same graph as published in the Dunne and Leng paper[31] though both have the same expected evaluation cost.

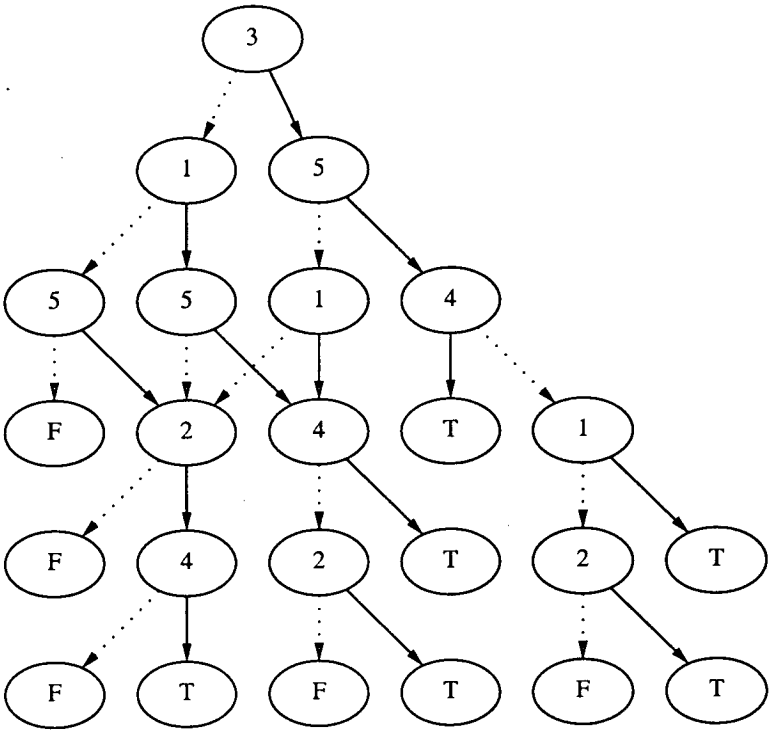


Figure 3.2: Minimum expected cost evaluation graph

3.4 Modes of operation

When a logical process has received a demand for its value and a calculation is required to satisfy that demand, there are a number of input and output modes in which the LP can operate.

3.4.1 Input modes

These modes define the way that a logical process can demand input values.

Single Mode: Each required input value is demanded separately and sequentially. The next demand is not sent until the previous demand has been

satisfied.

Broadcast Mode: This mode sends demands for all the input values that *may* be needed to calculate the result.

Parallel Mode: This mode sends demands for all the input values that *must* be needed to generate a result.

Group Mode: This mode sends demands for the minimum number of input values which *could* generate a result.

These different input modes are best illustrated with an example. Consider the threshold function T_3^5 as defined in Section 3.3.1, one of whose optimum call graphs is given in Figure 3.1.

Single mode would demand the value of input 1 and then, when the value arrived, it would follow the appropriate branch and then demand the next value. In this mode we descend the tree level by level until a fixed value is achieved.

Broadcast mode would demand all the input values at once. As the graph has 5 non-leaf levels, all 5 input values would be demanded. As they were returned their value would replace the node in the graph and incrementally the tree would be reduced to a fixed value. This may occur before all the results have been returned, e.g. 1=true, 5=true, 4=true. Subsequent input values can be discarded. The disadvantage of this mode is that, for non-strict functions, it requires computation to be performed that is superfluous to the final answer.

Parallel mode attempts to find all those inputs whose value must be known and then to demand their values in parallel. From the graph (Figure 3.1) we see that input 1 must be evaluated. We also see that, irrespective of the value of input 1, the next input that needs to be evaluated is input 5. Parallel mode would demand the values of inputs 1 and 5 in parallel. As each value was returned, the call graph would be reduced and, should it be clear that another input must now be evaluated it will be demanded in parallel with any existing, unsatisfied, demands. In our example, should node 5 evaluate to false no new demands can be made (the next input to be evaluated still depends on the value of input 1). Should input 1 evaluate to true then we can demand the value of input 3 and

input 2. We cannot demand the value for input 4 as whether this is needed or not is dependent on the values of inputs 2 and 3.

The value of a T_k^n threshold circuit can be determined by k true values or $(n - k + 1)$ false values. Let m be the lesser of k and $(n - k + 1)$. This is the minimum number of inputs which must be evaluated for the value of the function to be determined. In our example, m is 3. To determine which inputs to demand we firstly weight each mode in the call graph with the likelihood that it will be reached. For each input we sum the weights on all the nodes for the input. In group mode we request, in parallel, the m inputs with the largest likelihood of being called. In our example, those are inputs 1, 5 and 2.

The different input modes would be used to evaluate functions in the most efficient way. If, for example, the function was strict, then all the inputs require data and so broadcast is the most efficient. If, on the other hand, the function was non-, or partially-, strict, then a smaller number of data elements might be required. Single, parallel or group mode could then be more efficient.

3.4.2 Output modes

Just as there are different input modes, so there are different output modes. There is a complex interaction between the output mode of one logical process and the input mode of the logical process making the demand.

Demand-response: This mode only allows data to be sent out as a direct result of a demand for that data. This ensures that the data only reaches those logical processes that explicitly demanded the data.

Pre-emptive: This mode broadcasts the result to all the logical processes which might need the result.

Pre-emptive mode does not preclude the demand-response mode. The LPs which might need the result is based on some predictive model which attempts to predict future behaviour from observed past behaviour. A simple model would be to send the data to all LPs that have requested the previous data element. This model would be self-restricting, in that, should it accurately predict all those LPs

which needed the data before the LPs themselves knew that they needed the data, then no LPs would request the data and they would therefore not automatically receive the next data element.

Again, the use of different output modes would suit different conditions in the system. If, for example, a node might or might not send data to another node depending on conditions out-with its control, then that node should operate in request-respond mode. The action of sending a message to a node which does not need the data would waste memory. If it can be determined that a node will, at some point in the future, need the newly calculated result then the node should operate in pre-emptive mode. Doing this would avoid the overhead of a request and a reply. There is a potential for mixing the modes in any node. The node could, of course, ignore any pre-emptive data and then request it when needed.

3.5 Chapter Summary

This chapter looked at the benefits that are inherent in a demand driven system. It also addressed the more obvious shortcomings of the method and looked at ways to mitigate their effect. Threshold functions were introduced as a tool to address the question of strictness of a function. The work of Dunne and Leng[31] was introduced as an approach to minimising the work required to evaluate a function. This approach was then expanded to consider the evaluation of the function on a parallel machine through the use of input and output modes.

Having described the framework, the next chapter will describe the operation of two systems, one data driven and one demand driven, and will provide performance models for comparison.

Chapter 4

Performance Models

The previous chapter presented qualitative arguments on the merits of a demand-driven system. In this section we present three analytical models. These models capture the gross communication and computation behaviours of the ELSA [5], CMB[19] and general demand-driven systems.

This chapter first describes, in detail, the behaviour of the three systems under test. Then, after providing background definitions, models are derived which express the upper-bound of the gross computation and communication behaviour of those systems. The results of the models are then compared. The chapter closes with a discussion of some suggestions for improving the models.

4.1 The Conservative ELSA System

A system in ELSA is modelled as a weighted directed graph, where the nodes correspond to logical processes, the arcs to interconnections and the weights to the time delay on each arc. A few definitions follow:

- Any two nodes i and j in an acyclic directed graph $G(V, E)$ are connected, if the arc $(i, j) \in E$.
- If $\exists(i, j) \in E$, then (i, j) is an input arc to node j and an output arc from node i.
- $P_n \subset V$ is the subset of *primary* nodes, which have no arcs (i, n) in E.
- $T_n \subset V$ is the subset of *terminal* nodes, which have no arcs (n, i) in E.

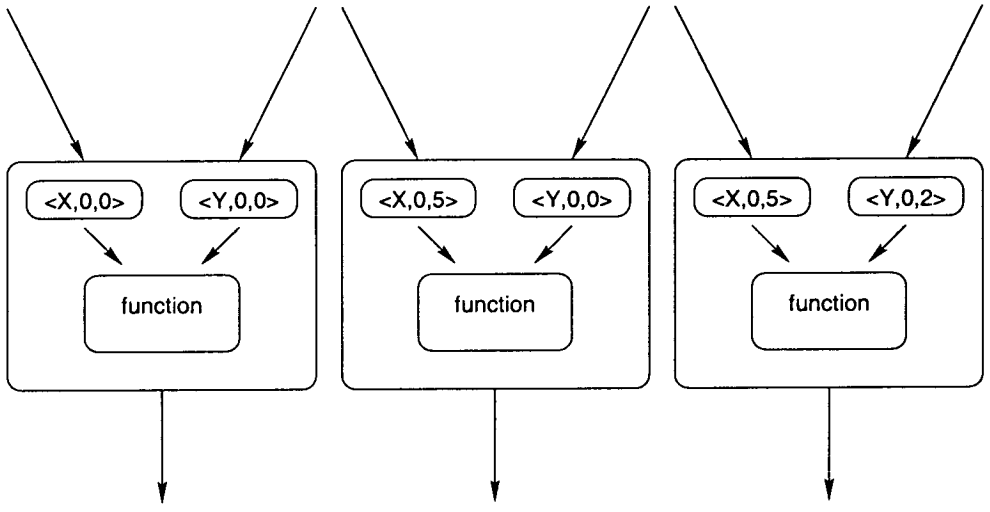
- The set $I = V - P - T$ of *internal* nodes.

Primary nodes place packets of information (or tuples) on arcs and terminal nodes remove them from arcs. An internal node places tuples on its output arcs, if and only if, it has removed a tuple from each of its input arcs. In its simplest form, a tuple has three fields: **V** – state field, **st** and **ed** – the start and end times for which the state field is valid. This has some similarities with the concept of look-ahead in that the interval could be considered as an event at time **st** with a look-ahead of **ed-st**. The difference is that while look-ahead states the minimum time for which the state is valid, the end time (**ed**) states the time at which the state becomes invalid. Associated with each input arc is a memory element which stores the tuple while it is valid.

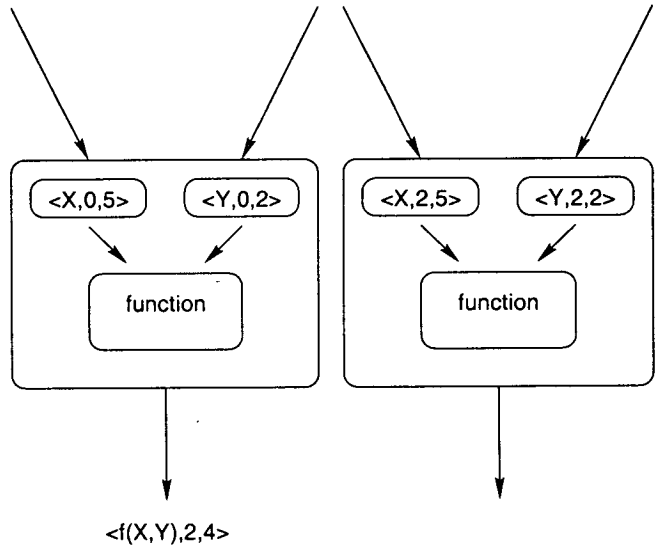
The system starts with an initialisation phase (Figure 4.1(a)). During this phase the input memory elements of all the logical processes have their start and end times both set to zero (the start of simulation time). The state need not be set as it will be overwritten by an incoming tuple before being read. Also, each logical process places a tuple on its output arc with the following information: state (V) is set to whatever value is desired, the start time (st) is set to zero and the end time set to δ, where δ is the simulation time taken for a change in an input value to affect the value of the output value (modelling the processing time in the simulated system). This tuple indicates the initial state on the arcs. It is safe to place this tuple on the arcs as the algorithm prevents any input to the node from affecting the output state until the node's delay δ has elapsed. This is justified as no event at the inputs can cause an event at the output requiring a start time less than δ.

Once this initialisation phase is complete, the nodes are ready to start. A node can only fire (generate an output event) when all its inputs have tuples whose start time is different from their end time. As all the arcs have been initialised with such tuples, as soon as they arrive at their destination, that node can fire. (Figure 4.1(b) and 4.1(c))

The firing of a node has two parts; the creation and sending of the output event (Figure 4.1(d), and the modification of the inputs to reflect the simulation



(a) Stage 1 (b) Stage 2 (c) Stage 3



(d) Stage 4 (e) Stage 5

Figure 4.1: A node in the ELSA system with $\delta = 2$. <V,st,ed> represents a tuple of state V over the interval [st,ed).

time advance (Figure 4.1(e)). The values for the output tuple are determined as follows:

- The state (**V**) is evaluated according to the functional description of the node.
- The start time (**st**) is the sum of the start time of the inputs (all inputs will have the same start time, this is shown later) and the delay δ of the node.
- The end time is the sum of the minimum of the end times and the delay δ. The minimum of the end times is used as that is the maximum time for which complete information about the input state is known.

Once the output tuple has been generated, the input memory elements can be updated. All the start times in the memory elements are set to the minimum of the end times, ensuring that at least one input requires to be updated before the node fires again. (This is how we can be sure that all the start times are the same when generating the output tuple.)

An internal node which removes tuples with consecutive time intervals from its inputs will maintain the sequence on its output. If the time sequence at the input starts at 0 and is consecutive (i.e. $ed_j = st_{j+1}$), then consecutive intervals will be maintained on every arc in the graph until the end-time of the last interval. The conservative ELSA algorithm is asynchronous and inherently deadlock free.

In cyclic networks, there is a potential for an explosion in the number of tuples. Consider the case of a node where its output is fed back to one of its inputs. In this case the start time for the new input tuple will differ from the start times for the other tuples by only the delay through the node. This will continue even if the state of the tuple does not change. Solutions to this, and other fragmentation problems are presented later.

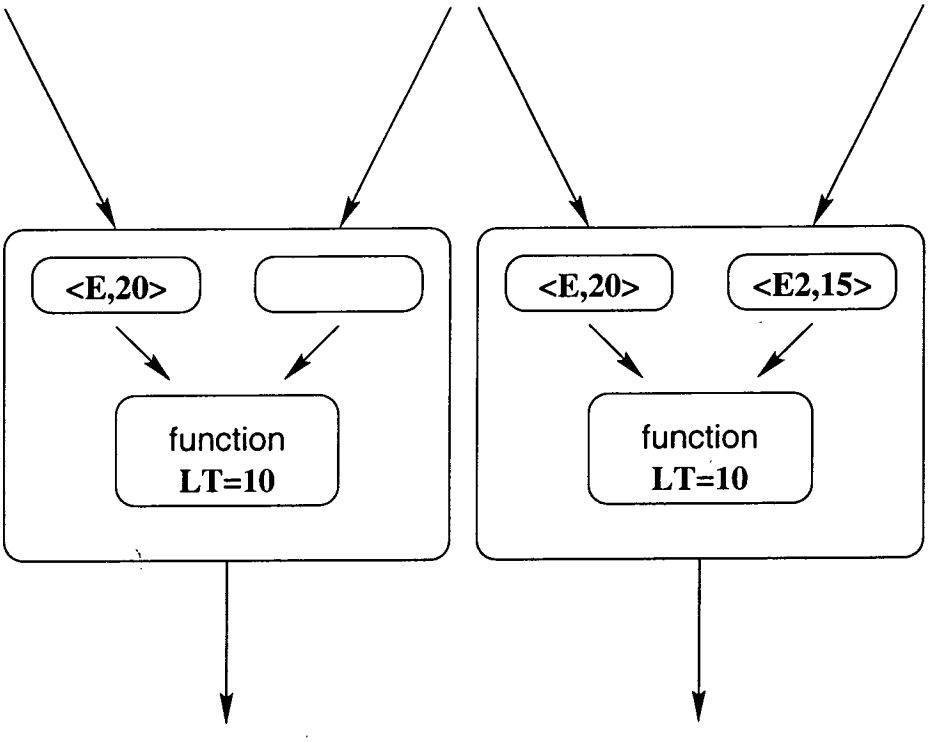
The handling of feedback is a well known problem for all conservative discrete event driven simulation[51].

4.2 The CMB System

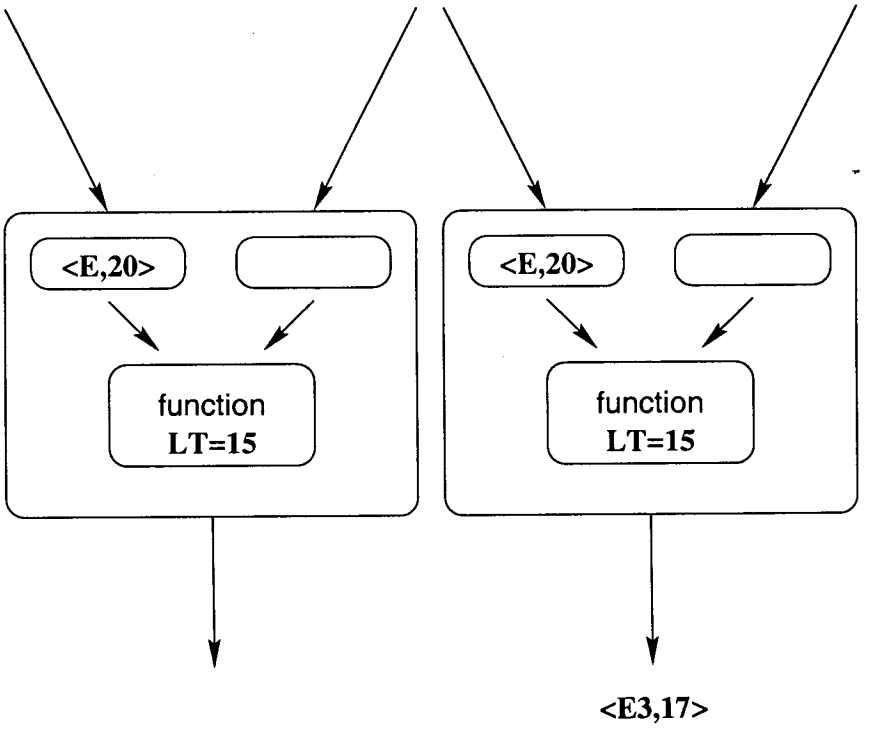
The CMB (Chandy-Misra-Bryant) system uses events which occur at an instant rather than intervals. These events indicate a *change* in state rather than the existence of a state as in ELSA. Associated with each event is a timestamp. When every input has an event pending, the logical process consumes the messages with the lowest timestamp. After consuming these messages, the LP advances its local clock and may send out one or more timestamped event messages.

As an example, consider a two input AND gate with a local time of 10 that has an event waiting on input 1 with a timestamp of 20 and no events pending on input 2 (Figure 4.2(a)). Thus we know the value of input 1 between times 10 and 20 (this, in effect, gives an implicit definition of the ELSA interval). While the AND gate is in this state, it must wait for an event message on input 2. Now suppose that the gate gets an event on input 2 with a timestamp of 15 (Figure 4.2(b)). The gate can now become active as it knows both inputs' states between 10 and 15. It consumes the event on input 2, advances its local time to 15 (Figure 4.2(c)) and possibly sends an output message with a timestamp of 15 plus the delay of the gate (Figure 4.2(d)).

In the basic CMB system, no messages are sent on an output line unless the value of that output changes. This optimisation, which is performed to make the simulation more efficient, is similar to that used in sequential event-driven simulators. However, in distributed simulation, this optimisation introduces deadlocks – points in time at which no LP can advance its local time because at least one input of every LP needs a message. For example, if the LP just described did not receive a message on input 2, it remains suspended. This deadlock is purely the result of the synchronisation mechanism and is unrelated to deadlocks in the physical system. This deadlock can be resolved in two ways: either by preventing it occurring in the first place, or by detecting and resolving the deadlock. The first solution is achieved by sending *NULL messages* whenever the local clock is updated but no output value is sent. This null message has the effect of propagating the local clock time to other LPs in the system. It is, in effect, saying that no message will arrive on this channel earlier than its timestamp. In the



(a) Stage 1 (b) Stage 2



(c) Stage 3 (d) Stage 4

Figure 4.2: A node in the Chandy-Misra-Bryant (CMB) system with $\delta = 2$. <E,st> represents a tuple of event E at the timestamp st.

second method the deadlock is resolved by scanning all the unprocessed events in the system, finding the minimum timestamp associated with these events, and updating the valid time of all inputs with no events to this time.

4.3 Demand-Driven Simulation

In contrast to the two systems described above, in which the computation is sparked by the existence of sufficient data on the inputs, the computation in a demand-driven system is sparked by the arrival of a request message at one of its outputs. The handling of the message could proceed in a number of ways. The simplest, most naïve, method will be outlined first followed by a system with a number of improvements.

When a request is received (Figure 4.3(a)), by the simplest system, a request is issued for the value of the first input (Figure 4.3(b)). When that value arrives (Figure 4.3(c)), if it is insufficient to determine the output value, a request is made for the value of the second input (Figure 4.3(d)). This continues until sufficient information is available at the inputs for the output to be calculated (Figure 4.3(e)). The newly calculated output value is sent to satisfy the request (Figure 4.3(f)).

There are two items of note which arise from the above description. The first is that more than one tuple might be generated to satisfy a single request. The second is that the tuples which satisfy a request need not arrive in any particular order. A demand-driven node handles these issues through the use of a calendar. A calendar starts with a single interval covering all simulation time, where the state of all the inputs is unknown. As each request is received, the node calendar is fragmented into intervals which are affected by the message and those which are not. The affected interval records the state change which has taken place. This might be a change from the node's value not being required to being required or it might be to record which input values have been requested and which have been received. Ultimately, the node's output state will be determined and this value will be placed in the calendar to satisfy subsequent requests without re-evaluating the node.

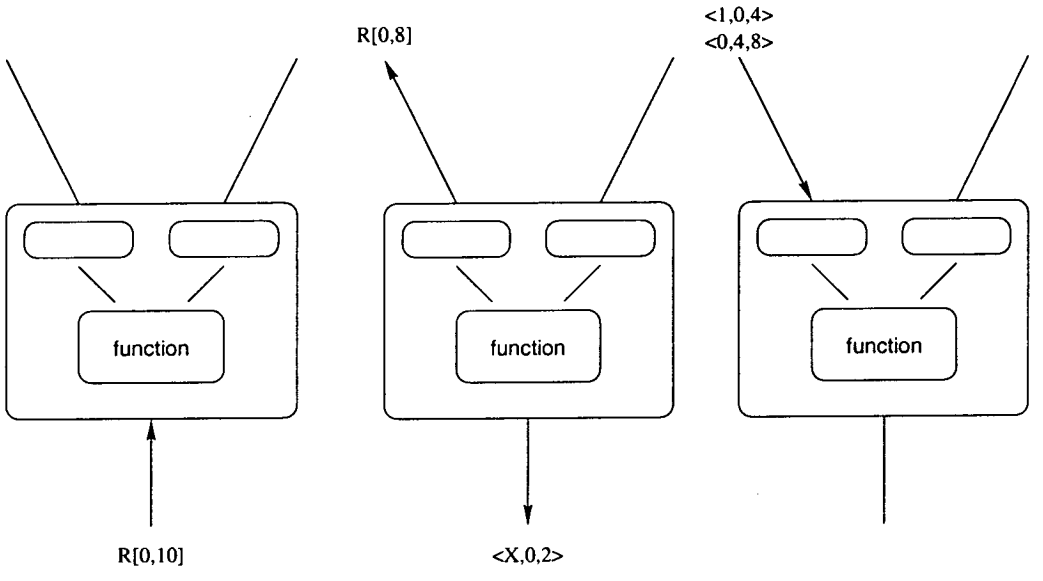
There are a number of problems with the system as presented. Firstly, each request received sparks the computation of the output value which would in turn spark multiple requests for input values. Secondly, though this does not affect the model, some functions require that a minimum number of input values be obtained before a result can be calculated. Requesting the members of this minimum group on an individual basis would slow the system and serialise the computation unnecessarily.

If every request was to result in a new evaluation being performed the number of requests would dramatically increase. Such an explosion in the number of request messages is one of the oft-cited reasons for not using demand driven evaluation. This can easily be overcome by the use of a memo facility which records the fact that a request has been issued by a particular input for a specific interval. The use of memo functions was mentioned earlier in the thesis,in Section3.1.1.3, with regard to reducing memory requirements. Now when a request is received, if a previous request has already started the evaluation for that interval, the request is stored and satisfied at the same time as the initial request. This system can easily be extended to store the calculated output state as well. Now, when a request is received, if it has been calculated before, it can be answered without generating any further requests. Such a system is reminiscent of the memo functions[43] used in functional programming languages. The advantage which their use in simulation has is that individual entries can be easily accessed as they are all indexed by the simulation time.

These stores or caches could grow quite large. It would be possible to prune the stores without causing any unit to recalculate any values. The simplest way would be to determine the latest time for which all the LPs had calculated values and delete any earlier entries. This is similar to the GVT calculation which precedes a fossil (garbage) collection in the Time Warp system.

4.4 Interval Manipulation

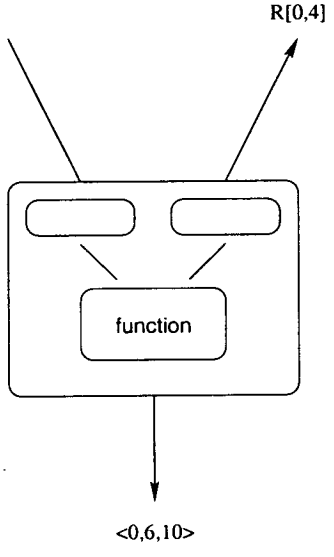
The systems with which we are working are based on intervals of time rather then the more common notion of instants. In the following sections we shall



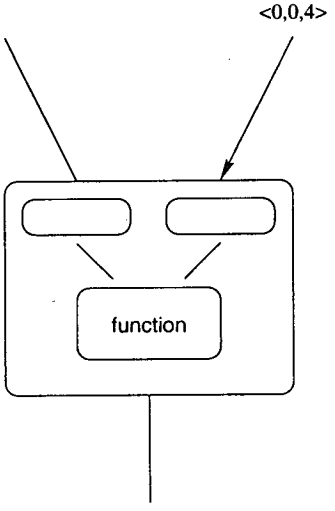
(a) Stage 1

(b) Stage 2

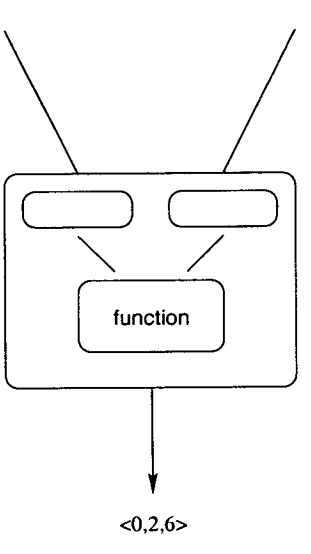
(c) Stage 3



(d) Stage 4



(e) Stage 5



(f) Stage 6

Figure 4.3: A node in the demand-driven system with $\delta = 2$. <V,st,ed> represents a tuple of state V over the interval [st,ed). R[st,ed] represents a request for the node's state over the interval [st,ed).

define intervals and relations and operations upon intervals. Using the interval system thus defined we will show how the ELSA and demand-driven systems would perform and thus how the analytical models of their behaviour were created.

4.4.1 Definition and relations

An interval $[a, b)$ covers all values, x, such that $a \leq x < b$. This definition rules out instants such as $[a, a)$ and intervals $[4, 2)$ where $a \geq b$. For interval t, we shall define the start time as t^- and the end time as t^+.

From the definition above it is obvious that

$$[a, b) \cup [b, c) \equiv [a, c) \tag{4.1}$$

Two intervals are said to overlap if, and only if, they have some interval in common.

$$\text{Overlap}(a, b) = \begin{cases} [max(a^-, b^-), min(a^+, b^+)) & \text{iff } max(a^-, b^-) < min(a^+, b^+) \\ \text{undefined} & \text{otherwise} \end{cases} \tag{4.2}$$

Ingalls[44] uses a similar definition but uses closed intervals. The use of closed intervals permits the existence of instants, i.e. [3,3], which are ruled out in our definition of an interval as we explicitly require a state to exist for a non-zero amount of time.

A **stream** is a set of intervals where no two intervals overlap. A **complete stream over** $[a, b)$ is a stream whose intervals are consecutive and which can, through repeated applications of the identity expressed in Equation 4.1, be shown to be equivalent to the interval $[a, b)$.

4.4.2 ELSA nodes

A strict ELSA node can only determine the output time interval when *all* the input intervals are known. The output interval is the interval which is common to all the input intervals. If we consider a two input node (inputs A and B) then the output intervals can be determined as follows:

1. Compare each interval in stream A with every interval in stream B to determine if the two intervals overlap (using the definition in Equation 4.2).

2. If there is an overlap, then add it to the output stream.

Assuming that A and B are complete streams then D will be a complete stream. In order to determine the output stream of an ELSA node, it is necessary to apply one more function. This function, δ, will move the stream forward in time to mimic the effect of the delay through the node. The δ function takes a time (t), which is the delay, and an interval (a). The δ functions adds the value of t to both the start and end times of the interval. For example, if the interval is [5,10) and the value of t is 3, then the result of the function will be the interval [8,13).

So far we have shown how to calculate the section of the output stream which is dependent on the input streams. The time between the start of the simulation and the first result appearing at the output of the node still needs to be accounted for. To do this we simple add another interval to the output stream. This is the interval [0,t), where t is the delay through the node. This is safe to do as the interval starting at time 0, the start of simulation time, will be advanced by the δ function, mentioned above, by t, and thus cannot affect the output over the interval [0,t).

It is therefore simple, if numerically taxing, to determine the number of intervals which will be sent down any given arc for an acyclic graph. Cyclic graphs add more complexity as their input streams must be merged with their output streams until a fixed point is reached but the basic operations are the same.

The above is a description of events which take place in an ELSA simulation. It only lacks data and function evaluation for it to be a complete description. While the above system will provide, not only the number of messages but the exact intervals, it also shows that the number of messages in an ELSA system is independent of the data being carried. This is not, as we shall see, true of either CMB or the proposed demand-driven systems.

4.5 Analytical Models

We assume that the system being modelled is represented by a directed graph, $G = (V, E)$, where V is the set of nodes and E is the set of arcs (i, j). On any one input arc all the events occur at discrete times and no two events on the same arc occur at the same time. The events which arise on any arc can therefore be represented as the set of times at which the event is raised. It is possible from the set of event instants to recreate the intervals used by ELSA. The output of any node is some function of its inputs at that time.

We are not directly concerned about when an event occurs, merely if an event occurs. The event set is therefore represented by the probability of an event occurring at time t.

We shall first outline the rules of probability before showing how to determine the amount of traffic in the system and then, finally, apply a work cost to calculate the work done by the system.

4.5.1 The Rules of Probability

Rule 1: If the probability of an event A is $p(A)$, then the probability that A does not happen is:

$$p(\text{not } A) = 1 - p(A)$$

Rule 2: Two events are mutually exclusive if they cannot both occur together. The probability that one *or* the other occurs is the sum of the separate probabilities:

$$p(A \text{ or } B) = p(A) + p(B)$$

Rule 3: Two events are independent if the outcome of the first has no effect on the outcome of the second. The probability that two independent events will occur is the product of the separate probabilities:

$$p(A \text{ and } B) = p(A) \times p(B)$$

Rule 4: This is an approximation that is often used in risk calculations. While it is not used in the simple models which will be used as examples, it would make

a sensible simplification for larger systems. Suppose that A and B are events, independent but not necessarily mutually exclusive, whose probabilities $p(A) = a$ and $p(B) = b$ are very small. What is the probability that at least one of them happens? It should be the sum of the following probabilities:

$$\begin{aligned} p(A \text{ and not } B) &= a(1-b) \\ p(\text{not } A \text{ and } B) &= (1-a)b \\ p(A \text{ and } B) &= ab \end{aligned}$$

Thus:

$$p(A \text{ or } B) = a(1-b) + (1-a)b + ab = a + b - ab$$

If a and b are small then ab is very small. So we can neglect the term $(-ab)$, and so:

$$p(A \text{ or } B) = a + b = p(A) + p(B)$$

In other words, events of small probability can be considered as being mutually exclusive, even if strictly speaking they are not independent of each other.

4.6 ELSA Model

In ELSA an output is generated whenever an event occurs at any of its inputs. If an event occurs at two or more inputs at the same instant then only one output is generated.

The input streams to an ELSA node are assumed to be independent. That is to say that the existence of an event on one input does not affect whether there will be an event on another input. This assumption only holds for tree circuits as every other structure will have at least one node with more than one output stream. The effect of this assumption on the validity of the model will be seen in a later example (Section 4.10.1).

In terms of the rules of probability given above, the transition function, T, for a two input gate is given below (a and b are the probabilities of an event on input A and B respectively).

$$T\ (a,b)\ =\ a\ +\ b\ -\ ab \tag{4.3}$$

The transition function for three inputs is $T(a, T(b,c)) \equiv T(T(a,b),c)$.

The probability of an event, p, multiplied by the run time of the system, n, gives a measure of the number of events which flow down the arc.

4.7 CMB model

The model for a CMB system using *NULL* messages is the same as that presented above for the ELSA system. This is because every message which arrives, whether it is a *NULL* message or a data message, causes the generation of a new message. This message is either an event message because the output value has changed or it is a *NULL* message sent to indicate an increased local time.

In the version of the CMB system which uses a deadlock detection and resolution mechanism, an output event is only generated when an output changes state. The model of the ELSA system is extended to allow for the probability that the output value may change. We denote this probability by E.

The transition function remains the same but the input parameters are now the product of the probability of an event, p, and the probability that that event is different from the previous event, E. This gives the probability of an event whose state is different from the last event.

The derivation of the value of E on an output is dependent upon the function which the node computes. Below we present the derivation assuming that an exclusive-OR function is being computed.

The nature of the exclusive-OR function is that if both inputs change state at the same time then the output will remain unchanged.

We therefore want the probability of either of the inputs changing state, *but not both.* This can be shown by the Venn diagram in Figure 4.4.

The shaded area of Figure 4.4 can be expressed as follows:

$$P_1E_1 + P_2E_2 - 2 \times P_1E_1P_2E_2$$

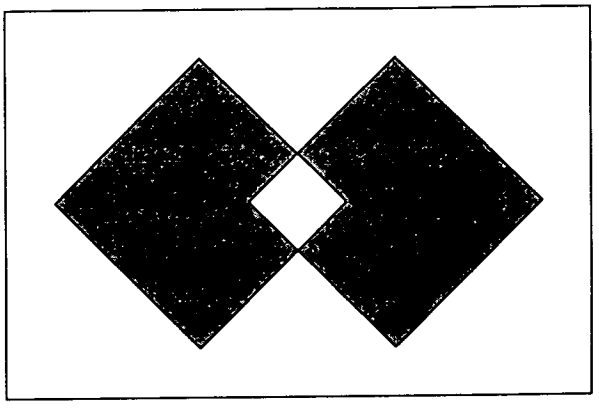


Figure 4.4: Venn diagram of A or B but not both

The probability of an event tends to the ratio of favourable events to trials, as the number of trial increases. The equation for the probability of an output event having a different state from the previous output is:

$$E = \frac{P_1E_1 + P_2E_2 - 2 \times P_1E_1P_2E_2}{P_1E_1 + P_2E_2 - P_1E_1P_2E_2}$$

As a new event is only dispatched along the output arcs if it is has a different value from the previous event dispatched along the arcs, we can use the value of E, defined above, to determine the number of real messages (as opposed to *NULL* messages) send along output arcs.

4.8 Demand-Driven Model

The demand-driven model is substantially more complex than either the ELSA or CMB models. This is due to the fact that the number of messages sent across any arc can be dependent upon the number of messages sent across some different arc. For example, the number of data messages being received on one arc can be affected by the number and state of the data message being received on another input.

The model is illustrated by a four port unit (Figure 4.5). Each port is bi-directional and is capable of both sending and receiving messages. This is required as each port must be able to send data messages and receive request messages or vice versa.

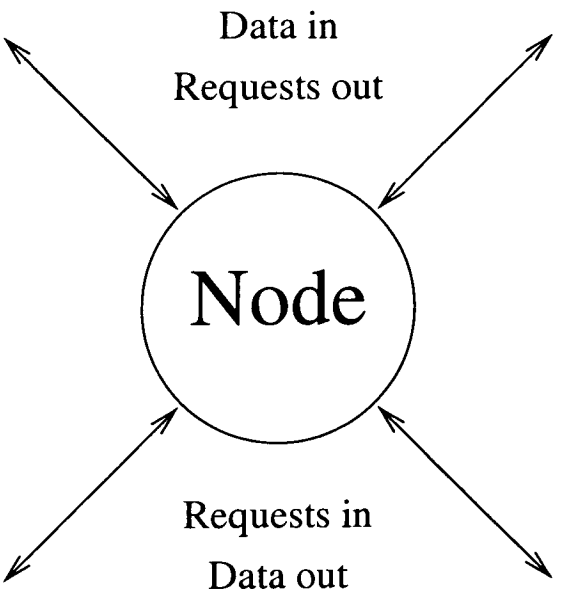


Figure 4.5: A sample node.

4.8.1 Communication costs

We make use of information already calculated for the ELSA model when we calculate the communications load for the demand driven approach. The ELSA model provides the communication load when the data on each link is required 100% of the time. With demand-driven simulation this is not the case. We shall use the ELSA calculations when we start to determine the demand-driven data communications load.

Each arc can be given a weighting (W) which will indicate the percentage of the time for which it is actually required. If a node has two or more output arcs which means, in turn, that the node has two or more sources of requests, then the effective amount of time for which the node is required to be active is some function of the percentages of the request streams. By the same argument used in Section 4.5.1, the function is T, the transition function.

The node will be active when one, or more, of the independent input arcs is active. Therefore, the effective fraction of the run for which the node is active is given by

$$W_i = T(X), X = \{W_{i,j} : (i,j) \in E\}$$

We now need to determine the weighting on each link in turn $W_{i,j}$.

If the node is active for W_i percent of the time then it must have, at least, one link active for that time. Therefore the first input will have requests covering W_i percent. For convenience we shall order the n inputs to a node and label them from 0 to $n-1$. We shall denote the source of the j^{th} arc as $src(j)$.

$$W_{i,src(0)} = W_i$$

The percentage of request activity on the second and subsequent arcs is dependent upon the function being evaluated at the node. There is a certain probability that more data will be required, for node i, once the first request is satisfied, let us call this $M_{(i,1)}$. The amount of request activity on the second input arc is $M_{(i,1)}$ times the activity percentage of the first arc. The value $M_{(i,2)}$ is the probability that more data is required after both the first and second arcs have been evaluated. This continues until all n of the input arcs have been evaluated. The value of $M_{(i,n)}$ must be zero as there are no further sources of data left to be interrogated. $M_{(i,0)}$ is 0 for all primary inputs (as they must, by definition, know their output without computation) and 1 otherwise.

$$W_{i,src(j)} = W_i \prod_{k=0}^{j-1} M_{(i,k)}$$

This enables us to estimate what percentage of the time each arc will be active.

Now that we know how many messages will be transmitted on all the arcs if they were active for all the simulation run, and the percentage of their activity, we can determine the number of data message passed over each arc.

$$DC_{i,j} = C_{i,j} \times W_{j,i}$$

There only remains the number of request messages sent over each link to be determined.

If there is only one output arc then the number of requests sent over the first input arc is the same as the number of requests received from that output arc. Should there be more than one output arc then the number of request messages is some function of the incoming request streams.

The function is, once again, the transition function T. The parameters of the function are based on the probability of a request message being received.

$$R_{i,src(0)} = T(X), X = \{R_{j,i} : (i,j) \in E\}$$

The model is completed with the generation of the probability of requests on the remaining input arcs. Each arc, apart from the first one, gets a number of requests based on the number of incoming messages on the previous arc. As mentioned earlier, each arc has associated with it a probability, M. The number of requests sent up the next input in turn is

$$R_{i,src(j)} = M_{(i,j-1)} \left(\frac{C_{src(j-1),i}}{n} \right)$$

The value $C_{src(j-1),i}$ is divided by n to get the probability of a data message arriving.

4.8.2 Computation costs

The handling of a request message should require significantly less real time than the handling of a data message. We denote by g the relative granularity of the work being performed at a node. A granularity of 10 means that the node takes 10 times as long to process a data message as it does to process a request message. Just as we noted in Section 3.1.1.1, data and request messages do not have the same communication or computation requirements.

To obtain a measure of work performed in a data-driven system we sum the work done at all of the nodes. The work performed at a node can be determined by the number of output messages which need to be generated and granularity. Note that this provides a relative measure and not an absolute statement of simulation time.

To obtain a measure of work performed in a demand-driven system we determine the useful work performed in the same manner as the data-driven system. We then need to add the number of request messages which need processing. Again, this gives a relative measure, but it can be used in comparisons.

4.9 Worked Example

In this section we apply the above model to a simple graph (Figure 4.6) to show how it is used in practice.

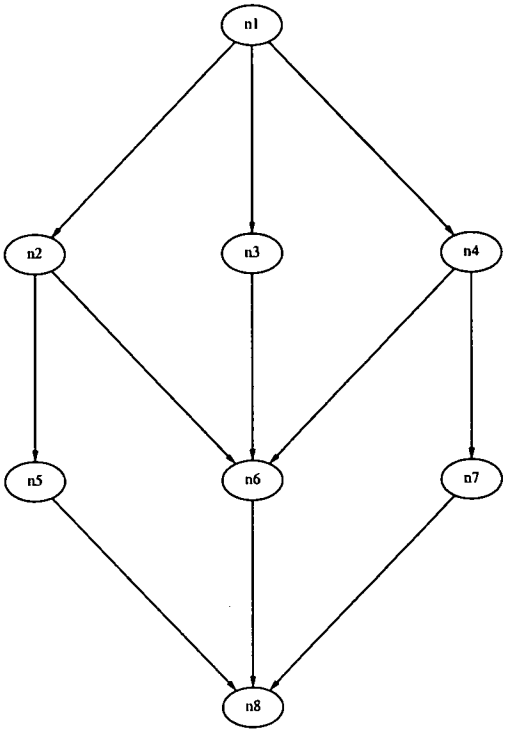


Figure 4.6: A simple acyclic directed graph

4.9.1 Summary of notation used

p The probability of an event.

n The run time of the system.

$C_{i,j}$ The data-driven communication load over arc (i, j).

E The probability that the event state is different from the previous event state.

$W_{i,j}$ The percentage of the total simulation time for which arc(i,j) is required to carry data messages.

W_i The percentage of the total simulation time for which node i is active.

$M_{(i,k)}$ The probability that more data will be required once request k of node i is satisfied.

$DC_{i,j}$ The number of data messages passed over demand-driven arc (i, j).

$R_{j,i}$ The probability of a request message on arc (j, i).

$T(a, b)$ The transition function which, given the probability of independent, but not mutually exclusive, events occuring on inputs a and b, can give the probability of an event occuring on input a, or on input b or on both a and b.

g The relative granularity of productive work to request message handling.

4.9.2 ELSA data-driven model

For the data driven model we shall assume that the two parameters p and n are known. This leads to equations for the number of dta messages on the outputs of nodes $1 \ldots 4$ as follows:

$$C_{1,2} = C_{1,3} = C_{1,4} = np$$

The data communication load from these nodes is the same as the data loads into the nodes, hence:

$$C_{2,5} = C_{2,6} = C_{3,6} = C_{4,6} = C_{4,7} = np$$

A similar situation holds true for the communication outputs from nodes 5 and 7.

$$C_{5,8} = C_{7,8} = np$$

The output of node 6 is a function of the inputs.

$$\begin{aligned} C_{6,8} &= T\left(T\left(\frac{C_{2,6}}{n}, \frac{C_{3,6}}{n}\right), \frac{C_{4,6}}{n}\right) n \\ &= \left(p^3 - 3p^2 + 3p\right) n \end{aligned}$$

This just leaves the data communications load for node 8.

$$\begin{aligned} C_8 &= T\left(T\left(\frac{C_{5,8}}{n}, \frac{C_{6,8}}{n}\right), \frac{C_{7,8}}{n}\right) n \\ &= \left(p^5 - 5p^4 + 10p^3 - 10p^2 + 5p\right) n \end{aligned}$$

Now that we have the communication loads for the ELSA data-driven model, we can turn our attention to the demand-driven case. Let us assume that we wish to know the output from node 8 for all simulation time. Therefore node 8 will need to request data for 100% of the time from node 5.

$$W_{5,8} = 1$$

As node 5 only has one input it must be active to fulfil all the requests. Therefore

$$W_{2,5} = W_{1,2} = 1$$

The percentage of time for which node 6 will be requested to be active depends on the function of node 8.

$$W_{6,8} = M_{5,8}$$

The percentage of time for which node 7 is active is again dependent on the function of node 8.

$$W_{7,8} = M_{5,8} M_{6,8}$$

Similar arguments can be applied to nodes 6 and 7.

$$W_{2,6} = W_{6,8} = M_{5,8}$$

$$W_{3,6} = W_{2,6} M_{2,6} = M_{5,8} M_{2,6}$$

$$W_{4,6} = M_{5,8} M_{2,6} M_{3,6}$$

$$W_{4,7} = W_{7,8} = M_{5,8} M_{6,8}$$

For nodes which have more than one output it is necessary to calculate the percentage of time for which they are active.

$$W_{1,2} = T(W_{2,5}, W_{2,6}) = 1$$

$$W_{1,4} = T(W_{4,6}, W_{4,7})$$

$$W_{1,3} = W_{6,3} = M_{5,8}M_{2,6}$$

The number of messages sent back down each arc is the total number of messages which would have been sent times the percentage of time for which the arc was active.

$$DC_{i,j} = C_{i,j}W_{i,j}$$

Therefore:

$$DC_{6,8} = C_{6,8}W_{6,8}$$

4.10 Verification of the Models

All three models were verified using the same circuit. It was a balanced binary tree with depth 11 (2047 nodes). Each internal node performed the same function which was exclusive-OR. The ELSA and CMB models were verified on a Breathing Time Buckets (BTB)[84] simulator written at DRA, Malvern. The demand-driven model was verified using the demand driven system described in the next chapter.

When the predicted output for the ELSA model was plotted against the actual output from the DRA simulator, it matched to within about $\pm$ 8%. Longer simulation runs reduced this figure further. The error is shown in Figure 4.7. Similar results were obtained for the other two models.

An important result from the model for an ELSA system is that given a sufficiently deep system then the output will increasingly tend to an event every time step. This is due entirely to interval fragmentation.

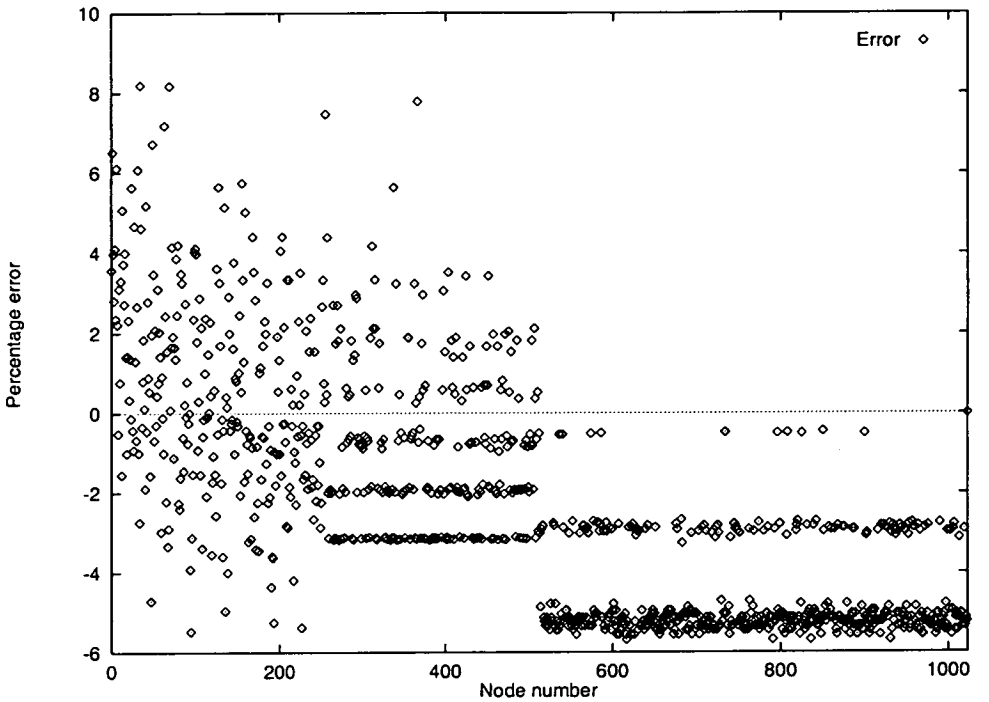


Figure 4.7: Percentage error when comparing ELSA model to actual results from DRA simulator.

4.10.1 The effect of non-independent streams

As mentioned earlier, the models assume that the streams are independent. A small case study using the gate level description of a 74LS283 adder[65] illustrates the limitations of the ELSA model. This limitation propagates through the other models as they are based, in part, on the ELSA model.

The parameters to the model were chosen arbitrarily:

- n, the run time, set to 1768.
- Values from the range $\frac{1}{80}, \frac{1}{85}, \ldots, \frac{1}{120}$ assigned to the nine input probabilities. No two inputs had the same probability.

The measured and calculated results are compared in Figure 4.8. Note that, while the model is a close representation in the early nodes (1...9) the quality of the model's predictions starts to decline in the next level (10...28) until, by level 3, the model is assigning over 5 times as many messages to an arc as actually pass across it. This over-estimate is the result of smaller over-estimates made in the levels above and the lack of independence in the data streams.

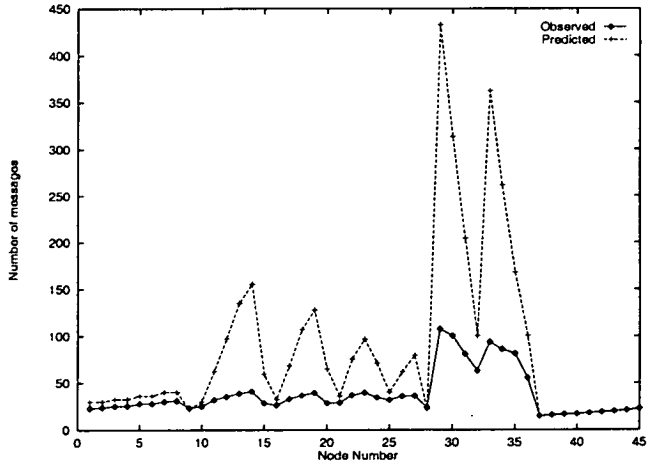


Figure 4.8: Comparison of calculated and observed communications for 74LS283 adder.

Consider the data stream {5,10,15,20}. Its size is clearly 4. If this stream is sent to both inputs of a two input gate then the output data stream, which is basically the union of the input data streams, will also have a size of 4. If however, one of the inputs had a delay of one unit then the output stream would be {5,6,10,11,15,16,20,21} which has size 8. It is therefore plain that the diverse paths taken from the data stream source to the destination gate can have a dramatic effect on the size of the output data stream.

Consider node 29 of the adder (a node on the 3rd level of the network). Its output stream is some function of its five input streams. The streams from nodes 10...14 are themselves functions of streams.

$$
\begin{aligned}
n_{29} &= f(n_{10}, n_{11}, n_{12}, n_{13}, n_{14}) \\
n_{10} &= f(n_2) \\
n_{11} &= f(n_1, n_4) \\
n_{12} &= f(n_1, n_3, n_6) \\
n_{13} &= f(n_1, n_3, n_5, n_8) \\
n_{14} &= f(n_1, n_3, n_5, n_7, n_9)
\end{aligned}
$$

As the above equations clearly show, the output from node n_1 is used four times in determining the output stream from node 29. The delays of the interven-

ing nodes are chosen randomly from the range [1...5]. With four nodes choosing from five discrete delays the probability that two or more of the nodes will choose the same value is quite high (0.808)[1]

When the range of possible delays in increased, the observed number of messages at node 29 (and at other nodes) increases as predicted. The increase, however, does not provide sufficient messages to meet the model's prediction.

These figures highlight the model's sensitivity to non-independent streams and to small errors high in the circuit snowballing and swamping the count at deeper levels.

4.10.2 Suggested improvements to the model

The greatest weakness of the model is that fact that it only considers local information in determining the data or request activity of a node. As we saw in the previous section, non-independent streams can cause the model to significantly over-estimate the number of messages on any arc and thus the total amount of work performed. This could be addressed in a number of different ways.

As we have seen in Section 4.4.2 it is possible, given the input intervals, to determine exactly the levels of traffic across arcs in the ELSA system. As all the models take as a base the data-driven level of traffic, then this might be used to produce a tighter upper-bound on the message traffic, and thus on the amount of work to be performed.

An alternative improvement would be to take into consideration the common sources of the message traffic and also the various paths from source to the node under consideration. If we have one stream taking two paths which had equal delays then we could remove from consideration one of the streams, as it gives rise to events which, as they occur at the same as those in the other stream, would not cause an increase in the number transmitted. Currently the model assumes independent streams and thus identical streams would be counted twice.

[1] The probability that two or more nodes will choose the same value is 1 minus the probability of all the nodes choosing different values. As there are 5^4 possible orderings of which $5\times4\times3\times2 = 120$ are unique, the number of orderings in which two or more nodes have the same delay is 0.808

```
int maketree (int st, int ed)
{
    int split;
    int left,right;

    if (st==ed) return (st);

    split = ((rand()%(ed-(st+1)))+1)|1;
    split += st;
    left = maketree(st,split-1);
    right = maketree(split+1,ed);
    SetFunction(left,right,split);
    return(split);
}
```

Figure 4.9: Algorithm to generate random binary trees

4.11 Tree Network Generation

The tree networks used to test the system are generated automatically. A simple algorithm randomly generates a binary tree with n nodes. The algorithm is shown in Figure 4.9.

Let $I_{(a,b)}$ be the interval of consecutive integers from a to b inclusive, i.e. $I_{(0,2)} = [0, 1, 2]$. Then $|I_{(a,b)}| = (b - a) + 1$.

The algorithm takes an interval $I_{(a,b)}$, such that $|I_{(a,b)}|$ is odd. The interval is split into three sub-intervals, $I_{(a,x-1)}, I_{(x,x)}, I_{(x+1,b)}$. The value of x is chosen randomly from the set $\{x : a < x < b \wedge |I_{(a,x-1)}| \; is \; odd\}$. The root of the tree is thus $I_{(x,x)}$, its two sub-trees are formed by recursively calling the algorithm on the intervals $I_{(a,x-1)}$ and $I_{(x+1,b)}$. The recursion terminates when it is called on an interval of size 1. The tree returned by such a call is a single leaf. This is shown in Figure 4.10.

4.11.1 Analysis of the Distribution of the Trees Generated

We note that the generation algorithm does not produce every potential tree with equal probability. To illustrate this we will first determine how to count the number of potential trees the algorithm will generate.

Let H_n be the set of *n-node* binary tree networks, and let $T_n = |H_n|$. Further,

let $H_{L,R}$ be the set of binary tree networks with L nodes in the left sub-tree and R nodes in the right. Let $T_{L,R} = |H_{L,R}|$. As the sub-trees on the left and right of the root node are independent the total number of possible trees is the product of the number of sub-trees on the left and right.

$$T_{L,R} = T_L T_R$$

Further, as the left and right sub-trees could be swapped, it is clear that $T_{L,R} \equiv T_{R,L}$.

In general, we have the following recurrence relation for T_n. It is defined in terms of the number of sub-trees generated by splitting the n nodes into sub-trees of x and $n-x-1$ nodes, where x ranges across the odd integers from 1 to $n-2$ inclusive.

$$T_n = \sum_{i=1}^{\frac{n-1}{2}} (T_{2i-1})(T_{n-2i})$$

The base case is $T_1 = 1$. This is a single node with no children or parents. As the number of nodes in a binary tree is always odd the next case is $T_3 = 1$. This represents a root with a leaf on both its sub-trees. The next case is T_5.

$$T_5 = T_{1,3} + T_{3,1} = T_1 T_3 + T_3 T_1 = 2$$

Values of T_n for small n are given in Table 4.1. It can be seen that T_n grows rapidly; for networks of only 29 nodes, there are over 2 million different potential trees that can be generated.

Now that we are able to count the number of trees with a given structure, we will show that the distribution of generated trees is non-uniform. Consider the generation of an 11 node tree. The number of tree with 11 nodes is given by:

$$T_{11} = T_{1,9} + T_{3,7} + T_{5,5} + T_{7,3} + T_{9,1}$$

The above algorithm would generate a tree corresponding to one of the terms above with equal probability of $\frac{1}{5}$. It should be noted, however, that $T_{1,9} = T_{9,1} = 14$ and $T_{3,7} = T_{7,3} = 5$ while $T_{5,5} = 4$. There are thus many more networks of type $H_{1,9}$ or $H_{9,1}$ than of the other possible types. In effect, the algorithm is biased towards producing balanced trees.

n	T_n
1	1
3	1
5	2
7	5
9	14
11	42
13	132
15	429
17	1430
19	4862
⋮	⋮
29	2674440

Table 4.1: Values of T_n for small n

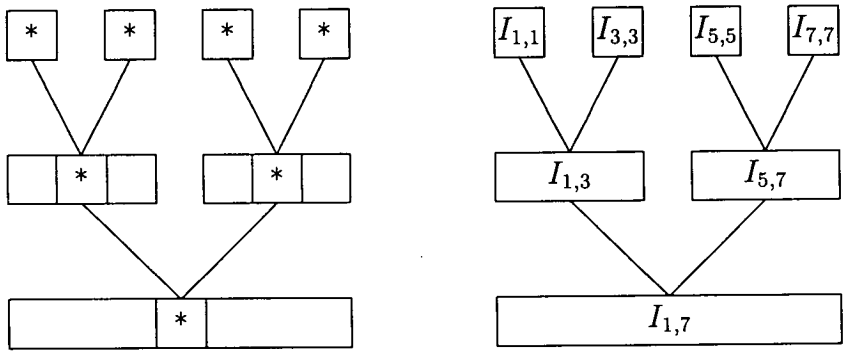


Figure 4.10: A decomposition of $I_{1,7}$

4.12 Results

Figure 4.11 shows the total communication load in data-driven and demand-driven systems for a 15-node balanced binary tree. The nodes were numbered sequentially, breadth-first, from the root. The fixed parameters for the model are: $p = 0.01$, M_2, M_3, M_4, M_6 and $M_7 = 0.5$, $g = 10$ and $n = 1000$. The values of $M_{(1,0)}$ and $M_{(5,0)}$ were varied in the range $0 \ldots 1$. The values of $M_{(1,1)}$ and $M_{(5,1)}$ were fixed at 0 as there remains no further data source to interrogate.

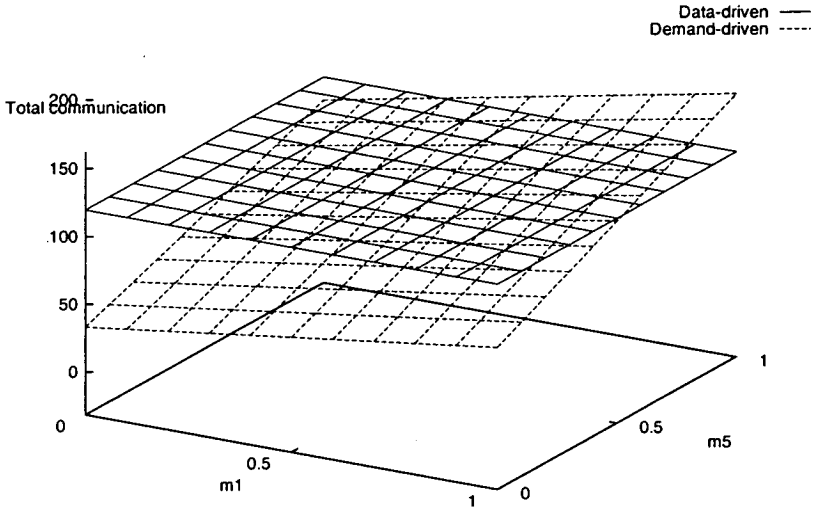


Figure 4.11: A comparison of data and demand-driven communication load for a 15 node balanced binary tree.

A Monte-Carlo analysis was performed on both the communication and work load equations. For the communication alone approximately 65% of the state space resulted in better performance for the demand-driven system. For work load alone approximately 95% of the state space resulted in better performance. As we can see, the amount of communication in the system is strongly related to the strictness of the functions at the nodes.

In order to assess the predicted relative merits of the demand-driven system in comparison to the data-driven a number of further analyses were performed.

4.12.1 Graphs

In contrast to the experiment above, the results presented in the rest of this chapter present the upper and lower bounds found after an exhaustive search of all possible fifteen node trees. Again, fixed values, except where explicity noted, were used: $p = 0.01$, $M_{1,\ldots,7} = 0.5$, $g = 10$ and $n = 1000$.

Figures 4.12 and 4.13 show the effect of increasing the granularity of the task at each node. As the granularity at the nodes increases the amount of communication remains unchanged. This is to be expected as the time taken at a node has no effect, in the model, on the number of messages transmitted. The total amount of work performed increases linearly with the increase in g (each node is taking longer to process data messages). It is worth noting that the total amount of processing resource consumed is consistently less for demand-driven. This is due to the effects of short-circuiting function evaluation.

Figures 4.14 and 4.15 show the effect of increasing the frequency of events at the inputs to the system. When $P = 1$ the data-driven systems are firing on every time step. As the frequency increases the amount of work increases more quickly for the data-driven system than for the demand-driven one. The amount of communication increases but appears to reach a point about $P = 0.85$ where both systems increase their communication at the same rate. The predicted effect on the amount of computation is that the demand-driven system will need to perform less work to achieve the same result as the data-driven system. The predicted effect on the amount of communication is that the data-driven system will need fewer messages than the demand-driven system. These two graphs indicate the slightly paradoxical nature of demand-driven systems in that they appear to consume more communications bandwidth to do less work than data-driven systems.

Figures 4.16 and 4.17 show the importance of non-strictness in determining which of the two systems is more efficient. Strictness has no effect on the amount of work to be performed by a data-driven system. We see that the point at which the maximum work from the demand-driven system equals the minimum work from the data driven system is just over a strictness of 0.85. What this means

is that even when node require both inputs 85% of the time, the demand-driven system is still predicted to require to consume less processor time. The effect of non-strictness is most notable on the communication load. As the strictness increases (more of the data is required to generate a result) there is a crossover point at about 0.65. Beyond 0.8, the demand-driven system is predicted to need more communiction bandwidth than the data-driven case. This break-even point could be increased by the use of pre-emptive data sending as this would potentially eliminate request messages.

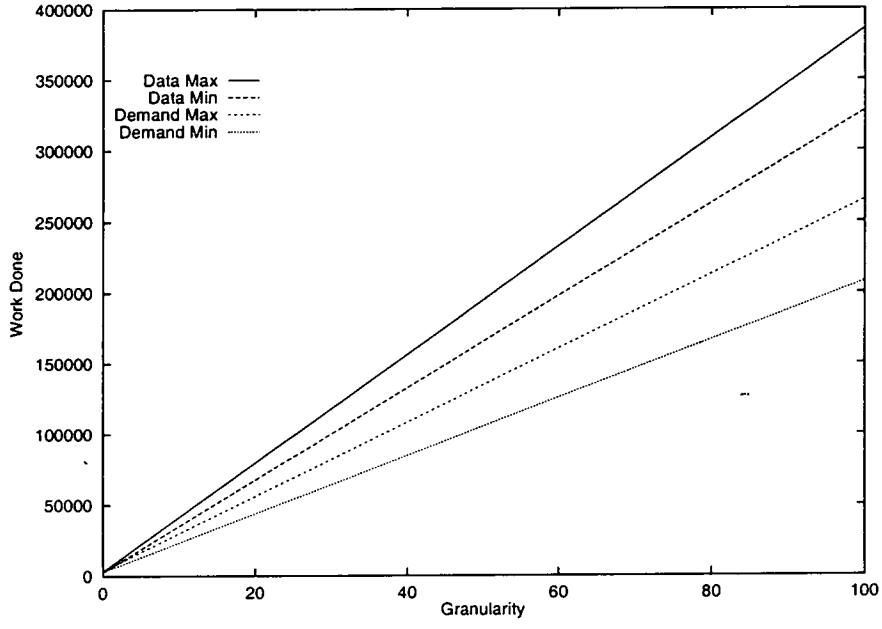


Figure 4.12: Effect of granularity on work done

4.13 Chapter Summary

In this chapter we described three simulation systems: ELSA [5], a data-driven interval based system, CMB[19], a data-driven event based system and our proposed demand-driven interval based system. Analytical models for the upper bound of the number of messages needed and the processing resource consumed were derived and some suggestions on how to make the upper bound more accurate were made.

The effect of the model's inability to handle non-independent streams effec-

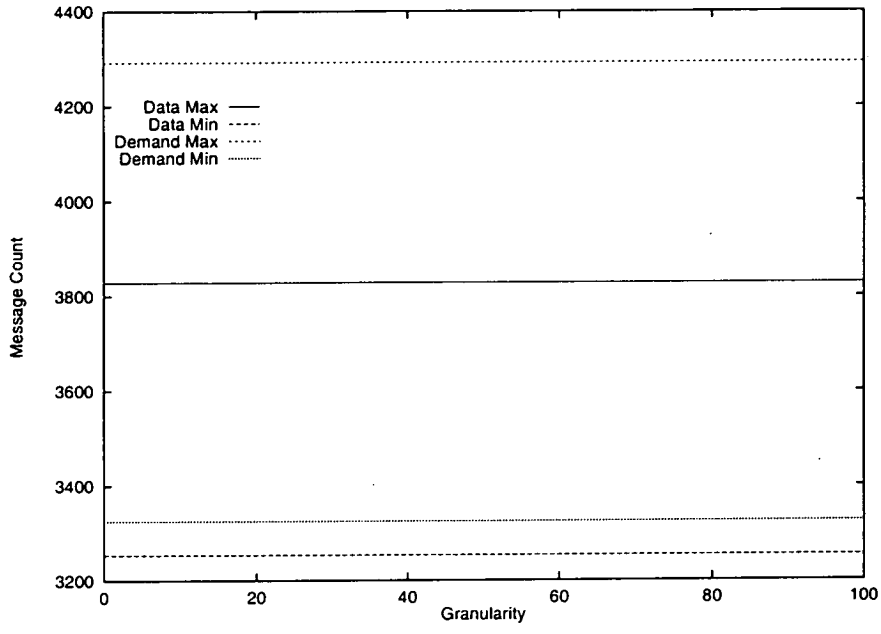


Figure 4.13: Effect of granularity on communication performed

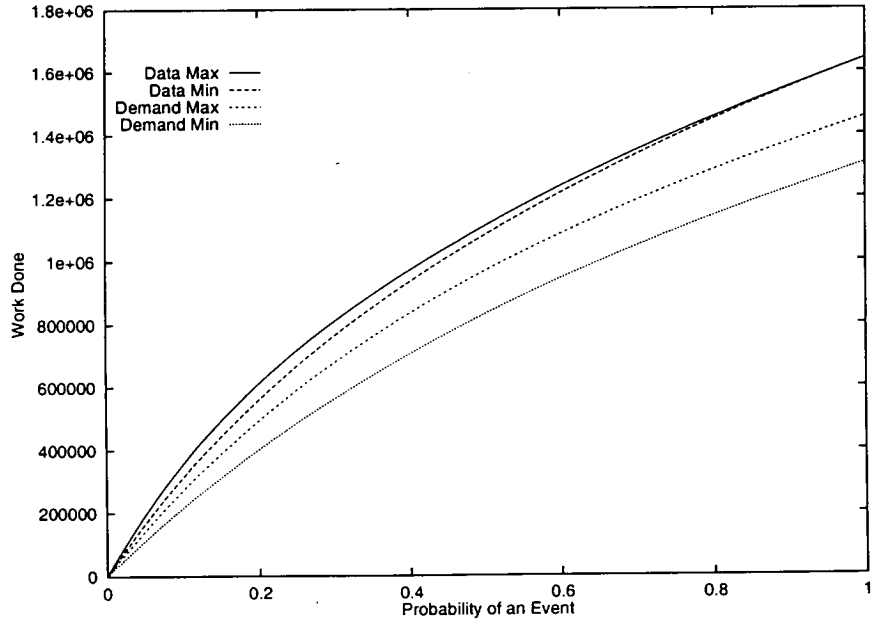


Figure 4.14: Effect of increasing the frequency of events on work done

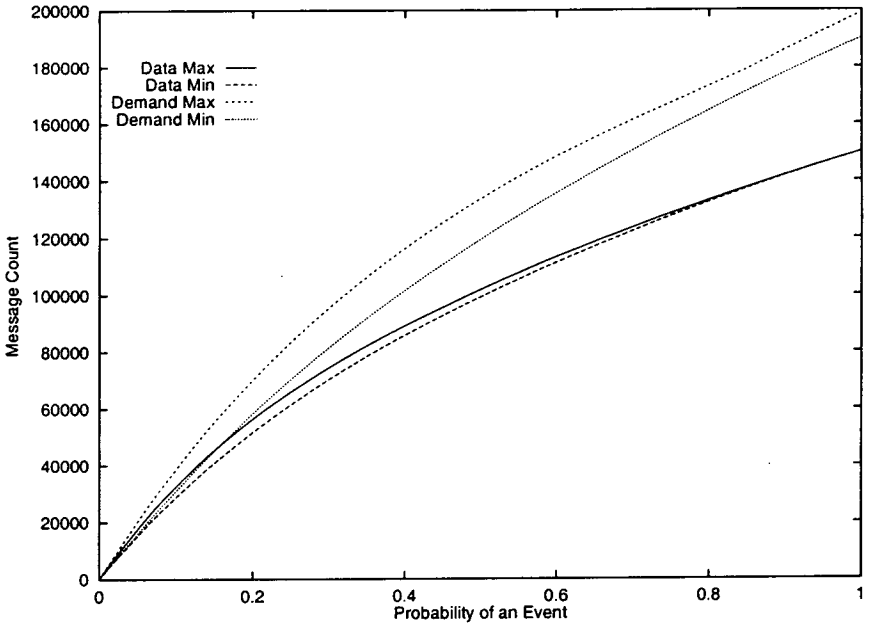


Figure 4.15: Effect of increasing the frequency of events on communication

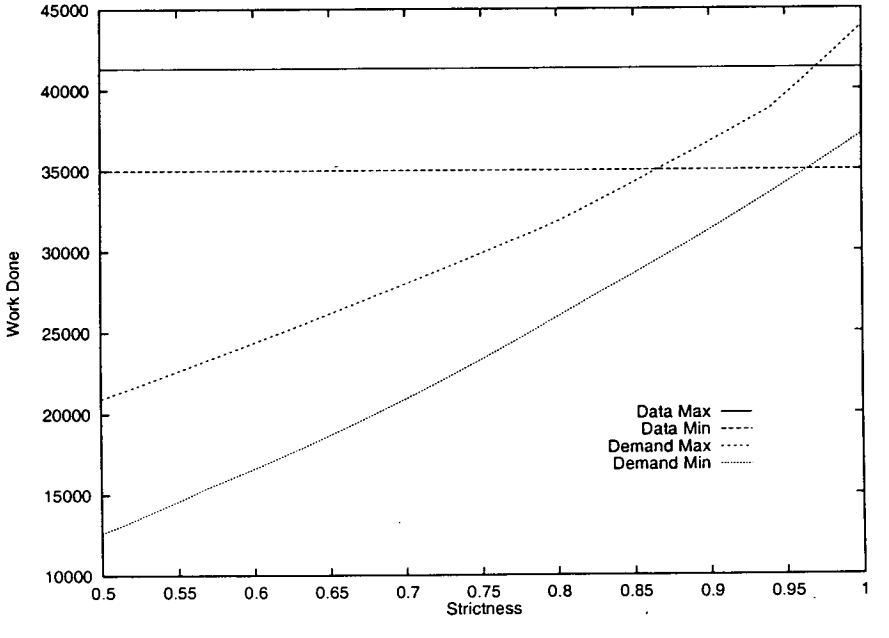


Figure 4.16: Effect of strictness on work done

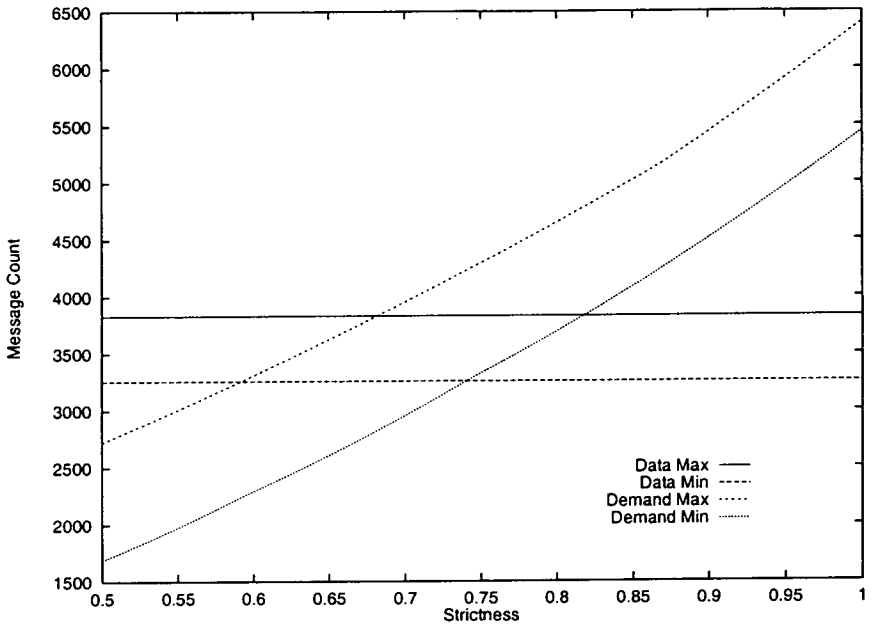


Figure 4.17: Effect of strictness on communication performed

tively was discussed. Random binary trees were generated and, as they exhibit independent data streams, were used to predict the expected performance of data-driven and demand-driven systems. The results of the models were presented in a number of graphs which show the effect of varying granularity, frequency of events and strictness on both the comunications bandwidth required and the processing resources consumed.

Chapter 5

Experimental Results

The model described previously is only able to give estimates of the total amount of computation or communication[1] performed by any node. It cannot provide information relating to the distribution of work through time nor can it give any indication of how increasing the resources can affect the performance.

In this chapter a number of different circuits are used to examine the dynamic nature of the simulation and, in particular, to focus on the parallelism and performance which is available as the computing resource increases.

5.1 The Test-bed

All the experimental results have been obtained from running the simulations within the controlled environment of a multiprocessor simulator. The reason for doing this was to have as much control of the "machine" as possible which enabled the results to be obtained without interference from other users or being dependent on factors such as caching or network load. The ultimate aim was to provide reproducible results. In order for these results to have any validity it is, of course, necessary to show that the simulator is a fair representation of a real system. This is shown later.

As the tests are performed within the test-bed it is necessary to define our concepts of simulation time and how it maps onto "real" time. Consider a circuit being simulated on the multiprocessor test bed. Simulation time relates to time as it is understood by the system being simulated – in this case, the circuit. System

[1] In future, the term **work** shall be used to mean either computation or communication.

time relates to the time of the test bed – this is the time used as real time.

The description of the test-bed has two parts: the micro model (how the processor functions) and the macro model (how the micro models are connected).

5.1.1 The Micro Model

The micro model describes a processor-memory pair. The memory is strictly local to the processor and there is no concept of global or shared memory. Nor is there any concept of a shared, or global, clock.

The micro model is designed as a reactive system. A reactive system is one where the units remain in some quiescent state until activated by a message or signal. Some computation is then performed followed by zero or more messages being sent to other units.

The messages which arrive at a processor are typed. The type determines whether the body of the message contains data to be processed or a demand for data[2]. The amount of time spent handling the message is dependent on its type and the number of intervals to which it is applied. The complete evaluation of a function for any time interval will require a number of messages to be processed.

When the node is idle (not processing any message) it waits until a message arrives and then starts to handle that mesage. Should another message arrive while the node is handling the first message, the arriving message waits in a queue until the processor is idle once more. Messages are handled in the order in which they arrive.

The handling of a message consists of updating the local state of the affected process (the processor can host a number of processes) and sending the resultant output messages, if any.

5.1.1.1 Message Handling

Each process maintains a separate state space which defines the state of that process throughout the simulation time. A process can be at different states in simulation time at any instant of system time.

[2]The number of such types could be increased to provide for a range of control messages. A use for such messages is presented later.

When a message is received the interval covered by the incoming message is extracted from the state space. Should the incoming interval start or end in the middle of an interval in the state space then that interval is split into two (the part not affected by the incoming message and the part which is affected). Once the affected portion of the state space is extracted the incoming message is then applied to each interval in turn. What action is performed depends on the state of extracted interval and the type of message received.

Demand Message:

When a demand message is applied to an interval, one of three actions can occur:

1. If the state of the output for that interval is known then the demand can be satisfied immediately. The output value is bundled into a message which is sent to the process which initiated the demand.

2. If the state of the output for that interval is not known, but data to calculate that state has been demanded, then the incoming demand is added to the list of currently outstanding demands. It will be satisfied as soon as the interval has a value.

3. If the state of the output for that interval is not known, and no earlier demands have been made, then the incoming demand is put in a list of outstanding demands and one or more demand is issued to the nodes whose data is required to calculate the value of that interval.

Data Message:

When a data message is applied to an interval, one of two actions can occur:

1. If there is sufficient data available for the value of the interval to be determined, then all the outstanding demands (and there must be at least one) are satisfied and the output value is stored.

2. Should there not yet be sufficient data available, then the incoming data value is applied to the function for that interval and a demand for further

data is sent to the appropriate node. The interval state is thus the partially evaluated result of the function. This enables short-circuit evaluation.

5.1.2 The Macro Model

The macro model describes how the final parallel resource is constructed and gives the characteristics of the communication links.

The communication graph assumes that any processor can send data to any other processor without interrupting the processing on a third, intermediate, processor. There are three times associated with communications. The first is the transport time; this is the time taken for a message to move through the network from the source to the destination. The second and third times relate to the processor work required to move the message to and from the processor into the communications net. If a message is being sent to another process on the same processor, this cost still applies.

Messages are queued at the destination in the order in which they arrive. Should two or more messages arrive at the same time, then they are queued in an arbitrary order.

In all of the experiments below, the nodes were scattered randomly across the processors. This was done to try and eliminate either method gaining an advantage from a more favourable distribution. For any individual experiment, both the data and the demand driven methods were tested using the same random distribution.

5.1.3 Test-bed Input/Output

Input and output is handled by special nodes which behave in a similar manner to all the other nodes being simulated. The input nodes can be considered as functions whose state is known for all simulation time, while the output nodes are functions with one input which merely store the incoming data.

5.1.4 Model Output

As the simulation system is itself being run in a controlled environment it is possible to take whatever measures are desired without affecting the system being studied: this is one reason for using such an enviroment.

The output trace concentrates on the behaviour of the processors[3]. The processor is constrained to be in one of four states and to start the simulation in the idle state. Whenever a processor changes state that information is written to the trace file and that state is known to persist until another state change event occurs. Extra information is written to the trace file depending upon what state the processor is entering. The exception to the above is the **Mark** event. This event is used to record any information which is deemed relevant but does not alter the state of the processor.

The states are given below:

idle: the processor is waiting for a message to arrive.

send: the processor is currently copying one or more messages onto the communications network.

recv: the processor is copying one message from the communications network.

task: the processor is occupied with internal processing and updating local system state.

5.2 Increasing confidence in the veracity of the simulator

As mentioned earlier, when using a simulator, it is necessary to obtain evidence that the simulator is, in fact, a reasonable representation of a real machine. The method which was used to obtain such supporting evidence is in three parts:

1. Obtain values from a real machine for the parameters of the simulator.

[3]The trace format is very similar to that used in PICL.

2. Obtain results for running the same circuit on both the real and the simulated machines.

3. Compare the two results

5.2.1 The gentle art of Ping-Pong

The time taken to send a message from one process to another can be measured by "bouncing" a message from one process off another process. By recording the time taken by the message to travel to the other process and back again, and assuming that the journey times are symmetrical, it is possible to determine the time taken for the message to travel half the distance. It is reasonable to suppose that the longer the message, then the more time it will take to transmit trough the network and, as such, the measures are taken for a range of message lengths. Two different graphs are presented below. The first (Figure 5.1) is for a multi-user machine (the specification of the machine is in Table 5.1).

Machine	Attributes	
Calvay	Make	Sun
	Model	SS10
	Memory	240M
	Purpose	Staff compute and Xterm server
Balta	Make	Sun
	Model	SS1+
	Memory	11M
	Purpose	Small desktop workstation

Table 5.1: Specification of test machines

Both graphs exhibit a similar linear trend. The graph for Calvay (Figure 5.1) also shows one of the problems inherent in trying to take performance measures on a multi-user machine, namely that the process is sharing the machine with many others, all of which are making demands on the processor. As such the times taken on Calvay can vary quite significantly, though, by taking sufficient results, a trend starts to appear. The graph for Balta (Figure 5.2) shows the linear trend much more clearly.

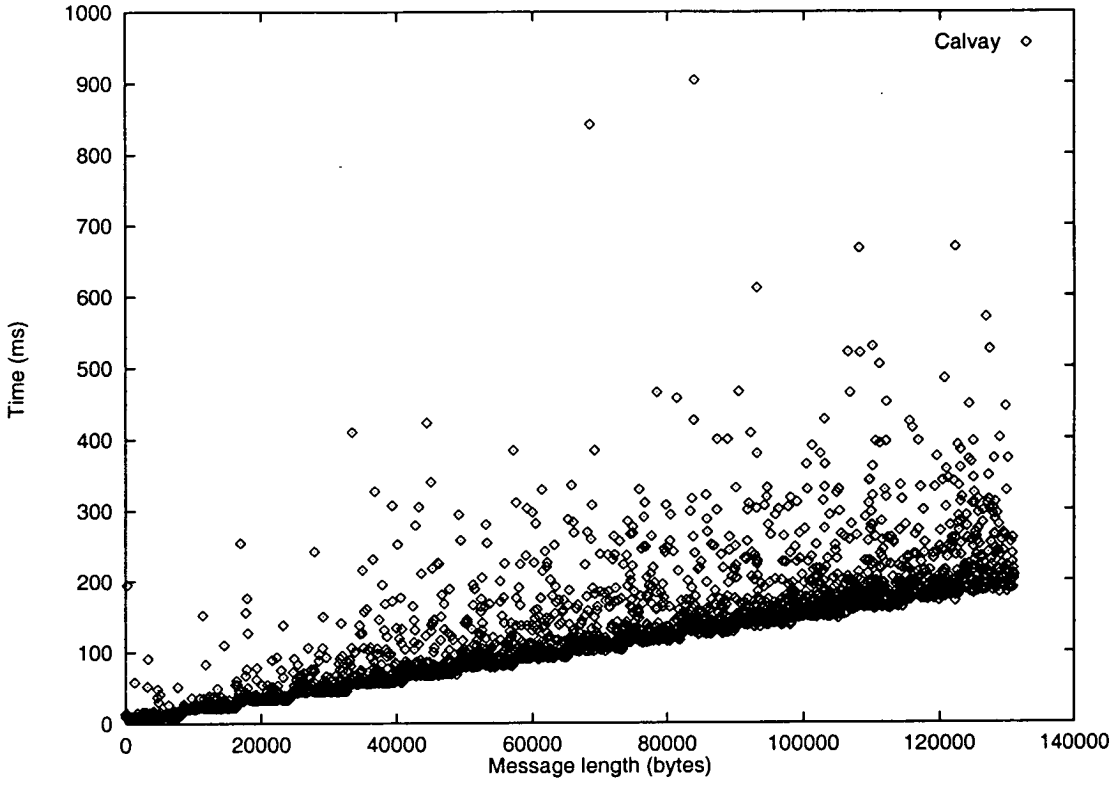


Figure 5.1: The time taken for a two way message on Calvay

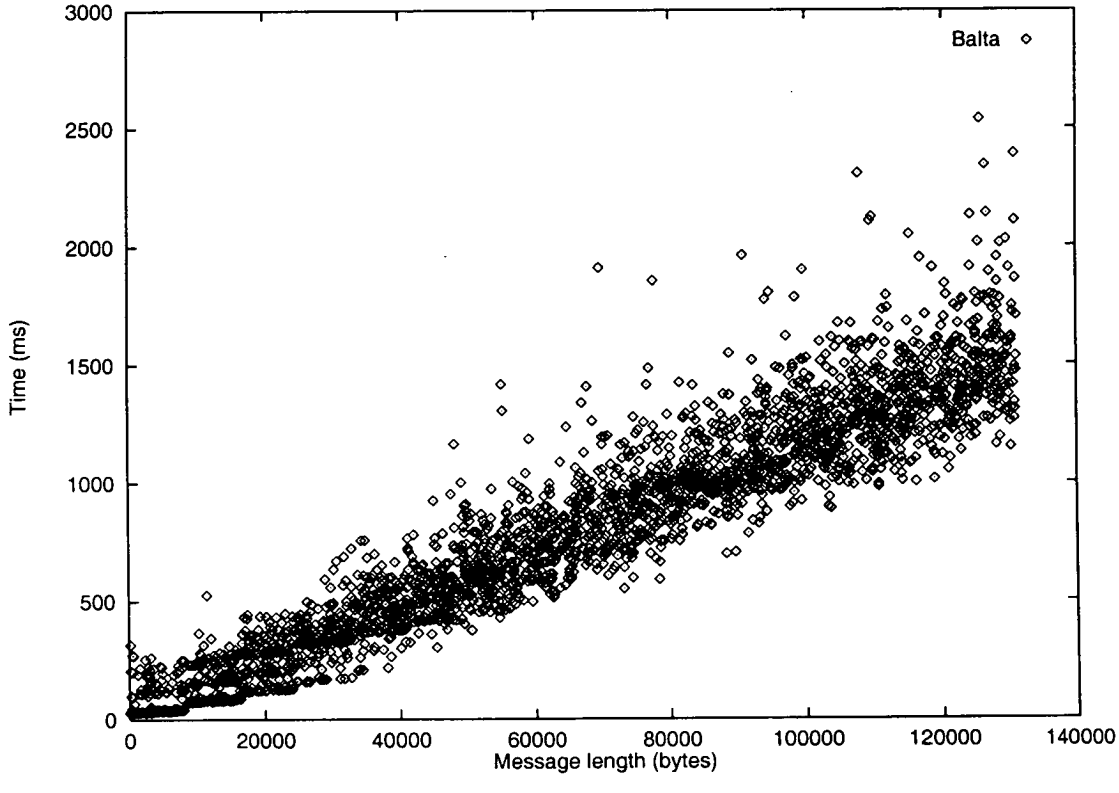


Figure 5.2: The time taken for a two way message on Balta

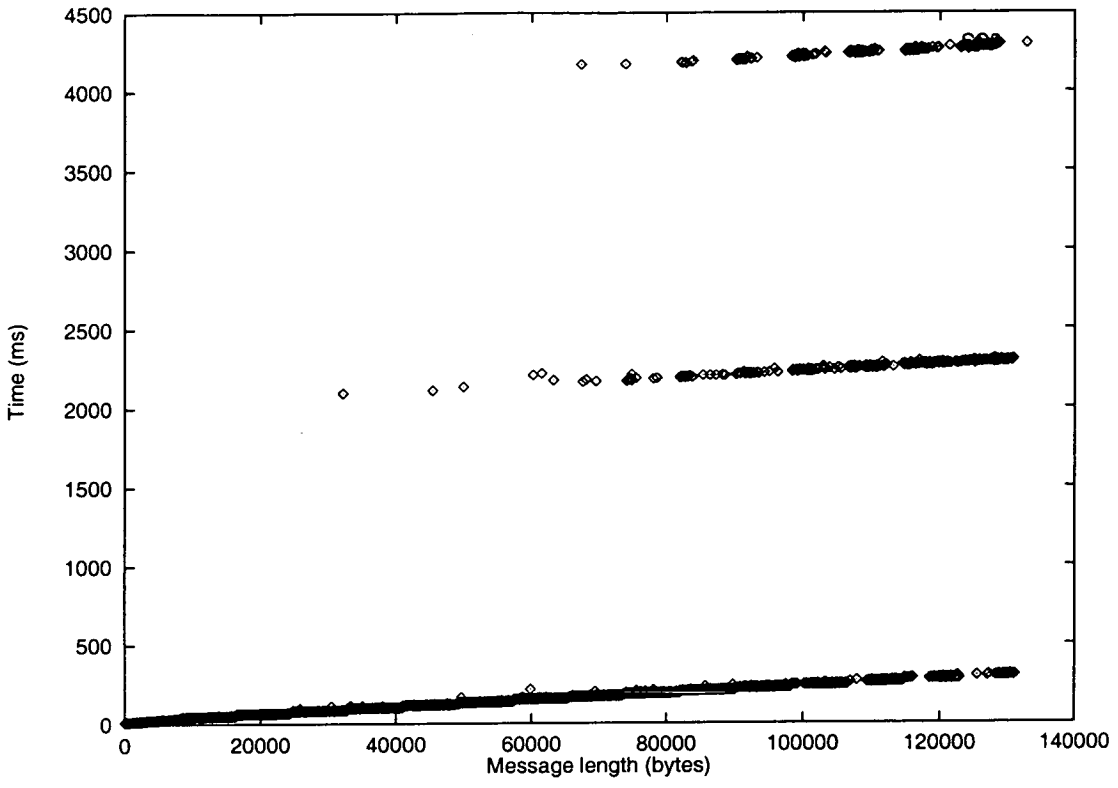


Figure 5.3: The time taken for a two way message on a pair of SS5 machines

The graph for a pair of SS5 machines is shown in Figure 5.3. This is the graph of the round trip times for a message being passed between the two machines. The graph shows some interesting features: firstly, that there is strong evidence of three separate bands of results and, secondly, that all three bands have a similar slope. This is shown in Table 5.3, which provides α and β values for the three clusters. The value of α is the intercept with the time axis while β is the slope of the data. Communication time is often modelled using the equation $\alpha+\beta l$, where α is the start-up cost, β is the per byte transport cost and l is the length of the message. The importance of these values is covered in Section 3.1.1.1. The three different α times reflect different start-up times for the communication. This may be due to the multi-user nature of the machines.

5.2.2 Time taken to handle Data and Demand messages

The time taken to handle either a data or a demand message was measured. A 32 node balanced binary tree was used to gather the results. Each node on the

Band	α	β
Upper	4001	0.0023
Middle	2031	0.0021
Lower	18.644	0.0022

Table 5.2: Values of α and β in milliseconds for two SS5 machines

Machine	α	β
Balta	24.597	0.0117
Calvay	7.6504	0.0018

Table 5.3: Values for α and β for two machines

tree was a two input logic gate. For each run of the system the total number of demand and data messages was counted and the total time required to handle each type of message was measured. The results shown in Figure 5.4 are of the average time taken to handle each type of message.

The average time to handle a demand message is 3.057 ms and to handle a data message is 2.167 ms. The reason that the time taken to handle a demand message is greater than that to handle a data message, in this case, is a combination of two aspects of the system. The first is that digital logic is a very fine grained computation and provides little overhead to the handling mechanism. The second is less obvious. When any message arrives it needs to update the calander of the node to say that a state has changed. For a demand message, this will frequently require a new entry to be placed in the calander. As the calendar has already been fragmented by a demand message, there is less chance of a data message having to fragment it further. The concept of a calendar was introduced in Section 4.3.

5.2.3 A comparison of the real and simulated systems

Both the real distributed simulator and the test-bed (simulated simulator) were set to simulate the same circuit. The test-bed was given the parameters measured from the real implementation and which are described above. The same circuit (255 node tree) was used and the average results of 10 runs are shown below (Figure 5.5). Each set of runs varied the number of processors from 1 to 10.

It is obvious from the graph that the real system is consistently slower than

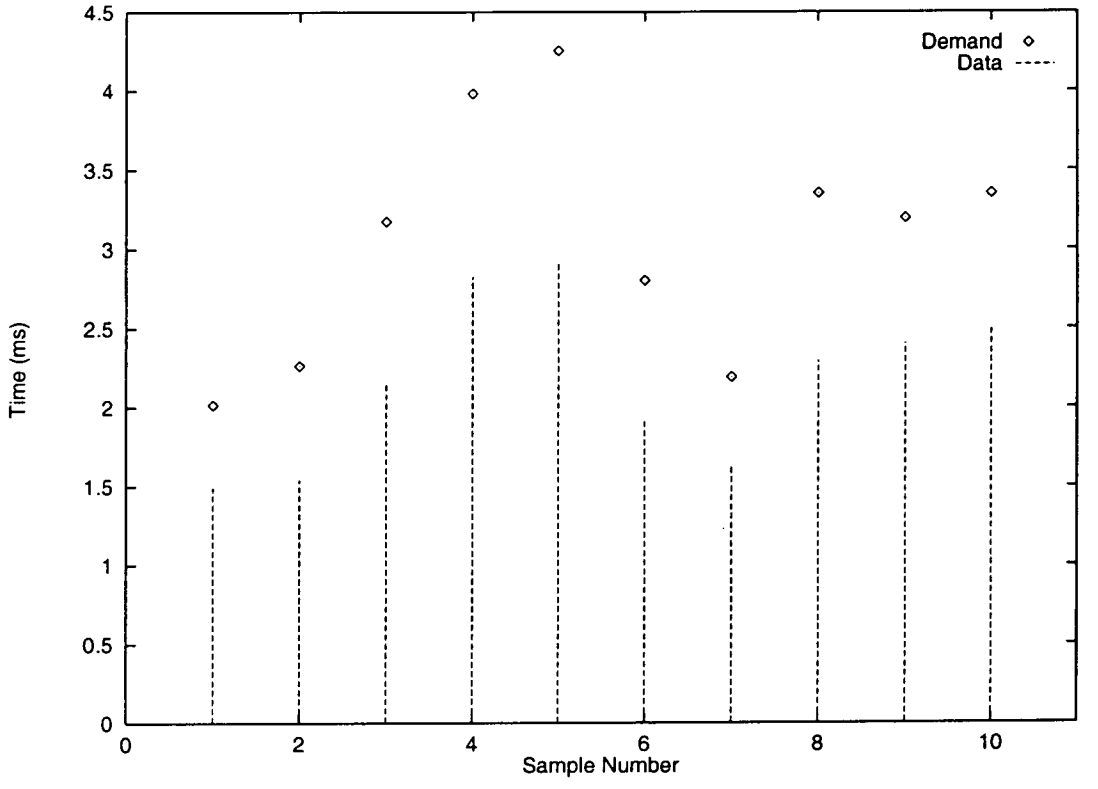


Figure 5.4: Samples of the time taken to handle a data or demand message

the test-bed (about 80% slower on average), but also that the test-bed does follow the same performance trend as exhibited by the real system, as the number of processors increases. The difference between real and test-bed simulators could have been caused by changes in the load on the network between the test-bed parameters being gathered and the comparison test being run.

5.3 The Measures

When undertaking performance measurement there is the question of exactly what should be recorded and how the collected data should be analysed.

The most common measure of performance is how long the system takes to produce the necessary results. This measure is particularly suited to high power parallel machines where the user is typically given complete and sole access to a number of processors. Until the user gets the result, all the assigned processors are unavailable to others, even if they are not performing useful work.

Another, increasingly popular, measure, is to calculate how much processor

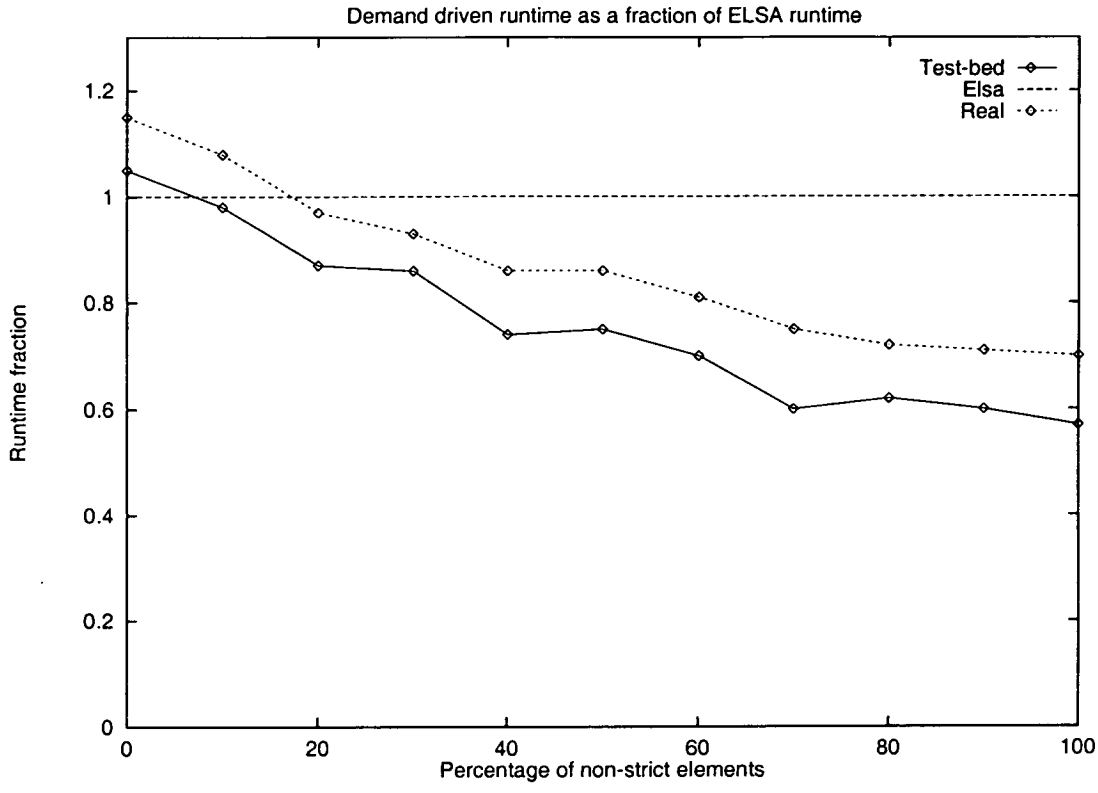


Figure 5.5: Runtimes of both the real and test-bed simulators

time is spent working on that particular problem. This measure is more suited to the Network Of Workstations (NOW) type of computing resource. In this senario the user has non-exclusive rights to a group of machines and shares the computing power with a number of other users. The other users may also be using the resource to run a parallel program.

Some measures are relevant to both types of computing resource: message count, for example, is important, as it helps to determine the load on the communication system. A point-to-point message count can indicate poor load balancing and potential hot-spots, which might slow the calculation.

5.4 The Circuits

A number of different circuits were used to test and compare the performance of data and demand driven evaluation.

5.4.1 Binary Tree

The binary tree circuit is shown in Figure 5.6. Each internal node of the tree is a two input threshold circuit. Each of the leaves of the tree is an input node. Each node in the tree has a unique path between it and the source. This implies that, if a node has enough information from a subset of its sub-tree to be able to fire, no demand will be made of nodes in any other sub-tree connected to that gate.

The graph shown in Figure 5.7 is for a 9-processor machine, but the trend is similar for all machines with fewer than 9 processors. The graph shows a two-dimensional result space. The lines on the graph indicate the contours in this performance landscape. The aim of the graphs is to give an indication of how the perfomance varies across various combinations of computation and commmunication times. The lines indicate points of equal performance. The closer the lines are together, the more quickly the performance changes. Lines parallel to the axis are unaffected by the change in value along that axis.

For data-driven evaluation it appears that the time taken to send messages between processors has no noticeable effect on the time taken to complete the evaluation. This would imply that the processes on the critical path are not idle awaiting the arrival of data. Demand-driven evaluation, on the other hand (Figure 5.8), does show the effect of increasing the time taken to send messages between processors. There is a distinct increase in completion time as the send time is increased, although it is only noticeable for low data handling times. Once the time taken to handle an incoming data message reaches 80 units, the effect of altering send times diminishes. This is due to there being sufficent work available so that individual processors are not starved of data.

Figure 5.9 shows the time taken to complete a simulation for both the data- and demand- driven methods. Also shown is the percentage difference between the two values. For the tree circuit, the demand-driven method consistently outperforms the data driven method by between 17 and 37 percent (averaging 26%). The particular results shown are for values of Tdata and Tsend of 640, but the results are similar to those obtained with other values.

Work time is a measure of how much work is performed by the processors and

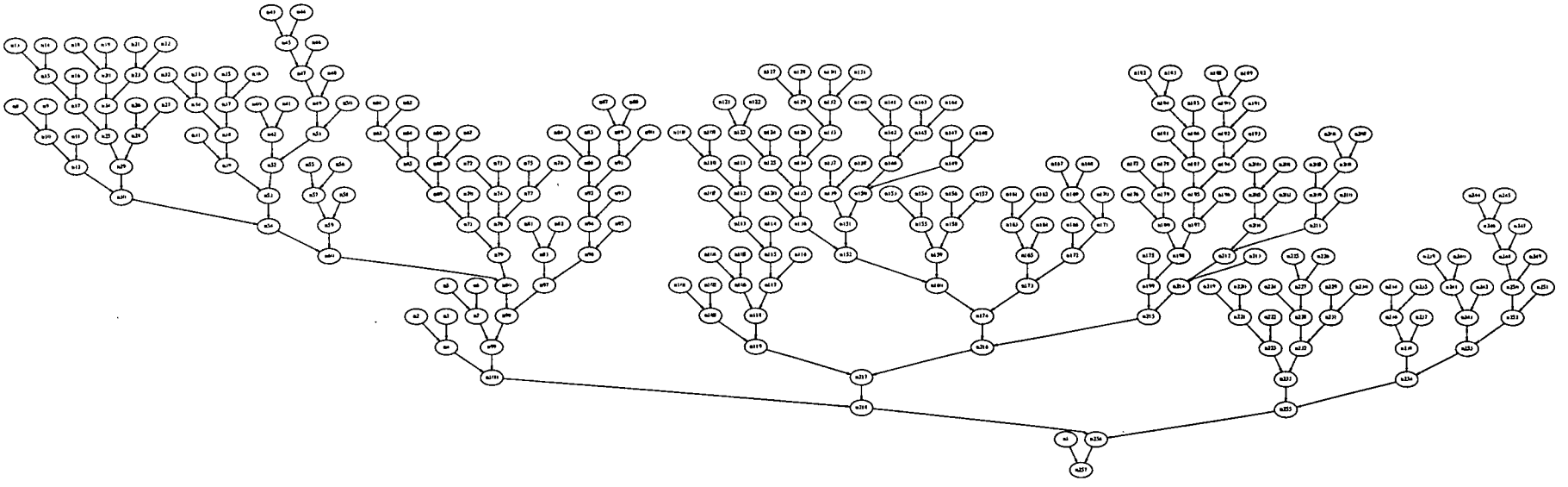


Figure 5.6: The 256 node binary tree used in the following experiments

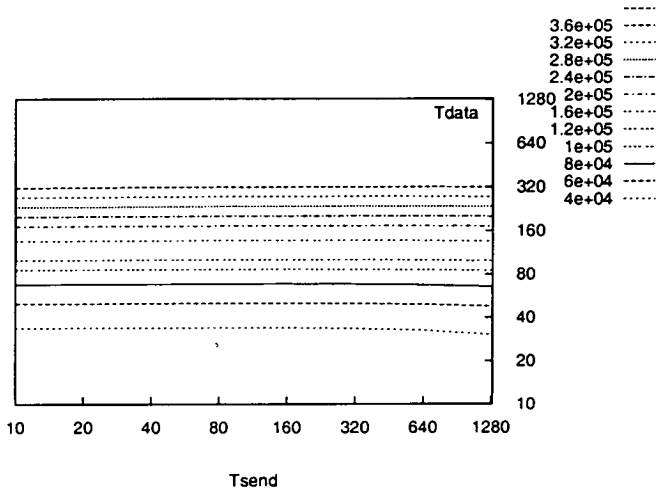


Figure 5.7: Tree 256: Data-driven runtime as a function of Tsend and Tdata for a 9-processor machine

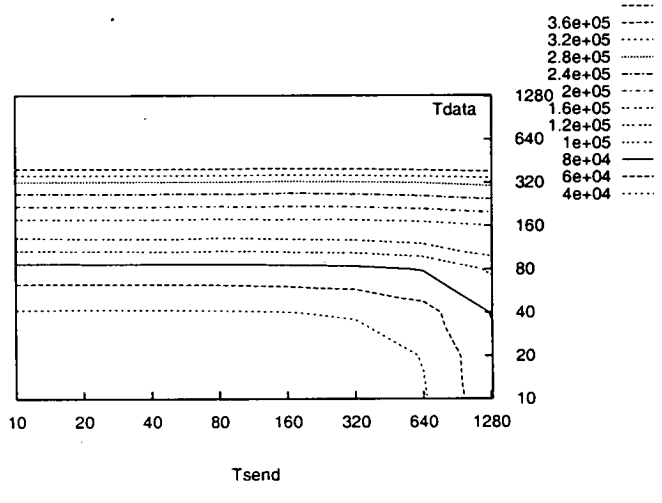


Figure 5.8: Tree 256: Demand-driven runtime as a function of Tsend and Tdata for a 9-processor machine

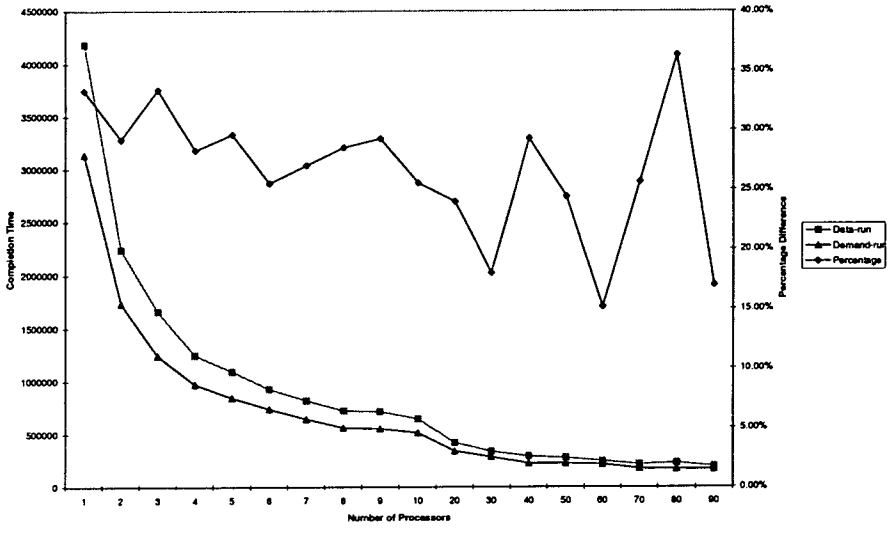


Figure 5.9: Completion time as a function of the number of processors (Note: the x axis is logarithmic)

as such is unaffected by variation in the time taken to send data from processor to processor. It is a useful measure in that it gives an indication of how evenly the computation load was spread across the machine and thus how efficiently the simulation uses computing resources.

Figures 5.10 and 5.11 show the traditional measures of speedup and efficiency. Both methods exhibit similar characteristics. Speedup and efficiency are related to the amount of elapsed time taken to complete the task. When we look at the graphs in Figures 5.12 and 5.13 we can see that, while the elapsed time for demand-driven (Rrun in the graphs) can be substantially more than that of data-driven (Drun), the amount of resource consumed by demand-driven simualtion (Rwork) is consistently smaller than data-driven (Dwork).

5.4.2 Adder

The adder circuit used is shown in Figure 5.14 and is the gate level description of the 74LS283 adder circuit[65]. The adder performs the addition of two 4-bit binary numbers. The sum outputs are provided for each bit and the resultant

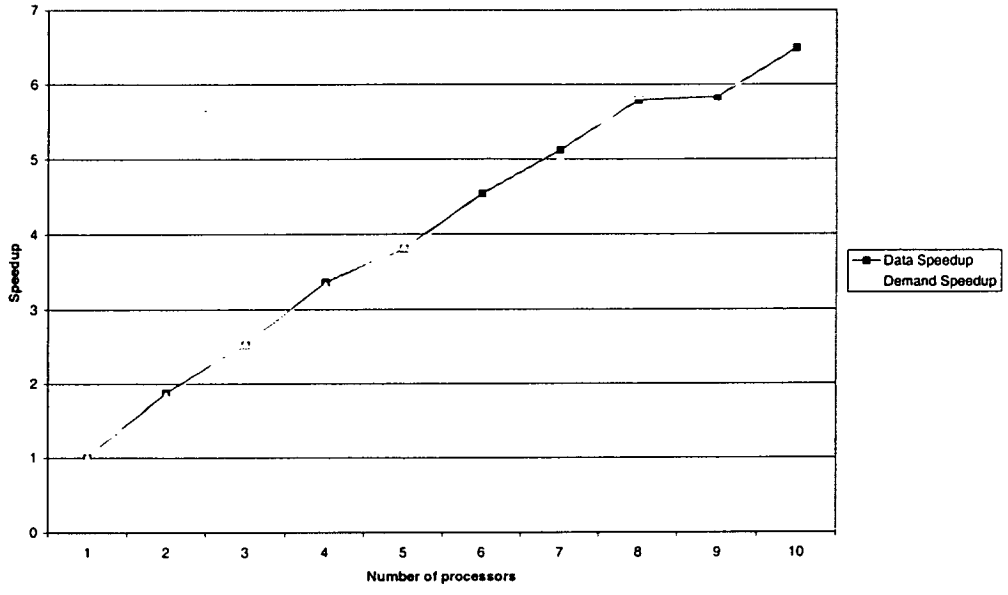


Figure 5.10: Speedup evident in a 256-node tree for both data- and demand-driven simulation.

carry is obtained from the fourth bit. The adder features full internal look-ahead across all four bits, which provides the system designer with partial look-ahead performance at the economy of a ripple-count implementation.

Figures 5.15 and 5.16 show the completion times for the adder on a 9-processor machine as the values of Tsend and Tdata are varied. Both graphs show a similar behaviour with the demand-driven system being slightly slower for all values. There is, in the demand-driven system, a very slight curve at the end of the division between the first and second ranges. This implies that, for large values of Tsend, the time taken to compute the result, Tdata, becomes significant. The Tdata value would only have an effect on the run time if nodes were having to wait for data to be produced, as, otherwise, the computation time would be covered by the communication time. The slight downturn is in marked contrast to the result from the tree (Figure 5.8) where the curve is significant and starts to have an effect at much lower values of Tsend.

The four graphs (Figure 5.17–5.18) display the run times and amount of

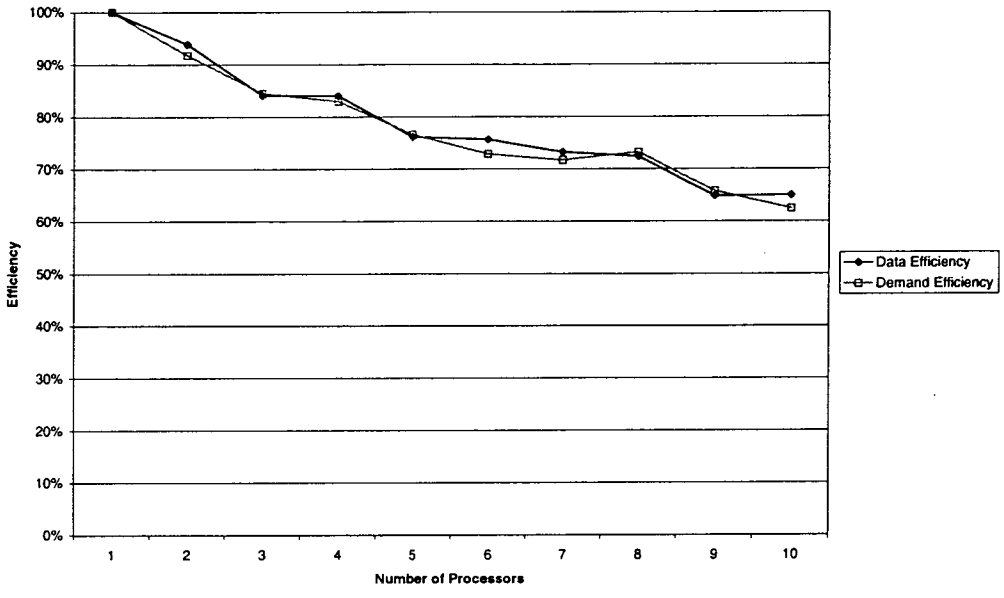
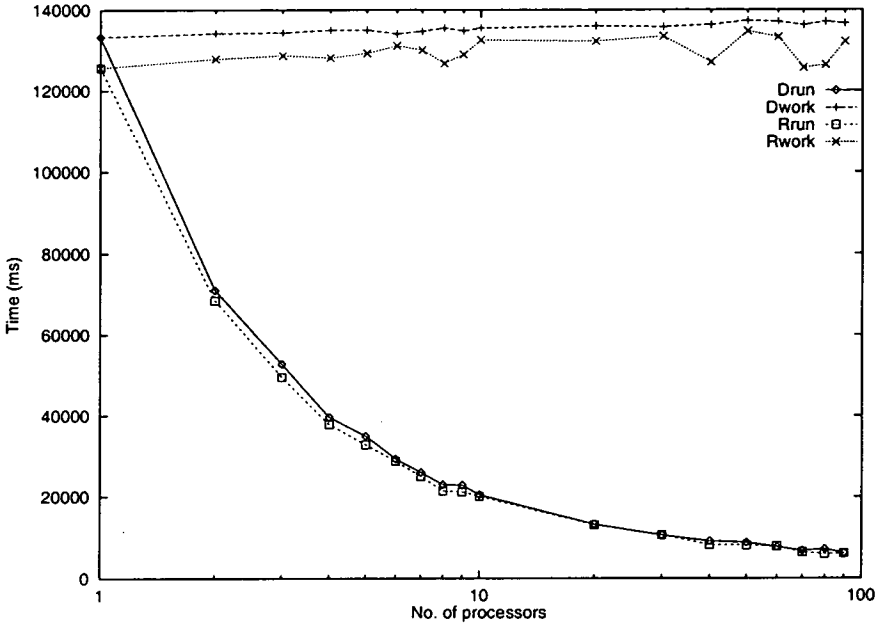
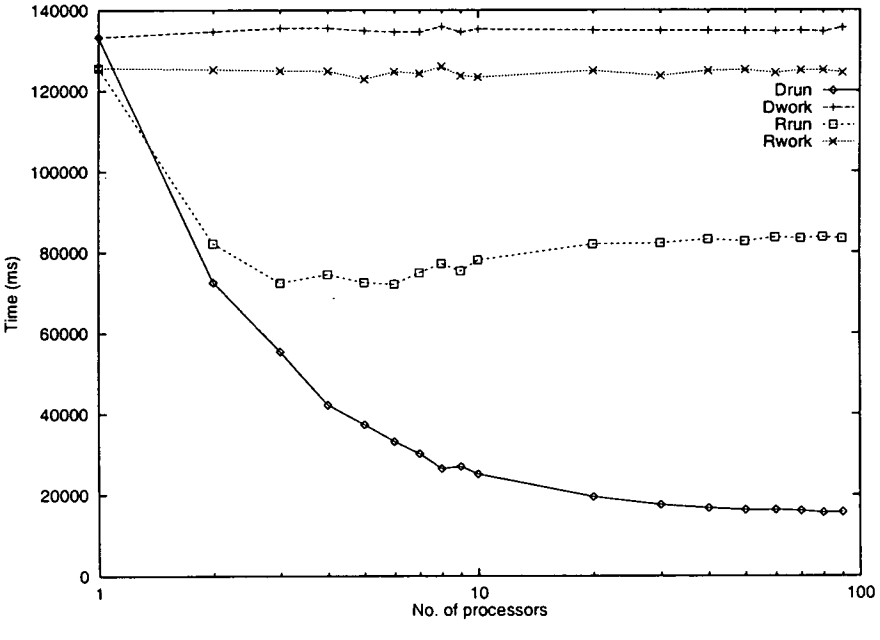


Figure 5.11: Efficiency evident in a 256-node tree for both data- and demand-driven simulation.

work performed for particular values of Tsend and Tdata for a range of machine sizes. The most important result is that the data-driven system consistently outperforms the demand-driven system. The second result is the sensitivity to the value of Tdata which is exhibited by the demand-driven system. Comparing the graphs in Figure 5.17(a) and Figure 5.18(a), the runtime values for the data driven system are little changed. The demand-driven system is, however, dramatically affected for Tdata=1280; not only are the completion times much higher, but the system fails to make any performance improvement after 3 processors. The resaon behind these results is the fact that the successful evaluation of many of the nodes is dependent on the evaluation of two nodes (nodes 3 and 5 in Figure 5.14). These nodes have a large fan-out and thus the processing becomes serialised on these nodes.

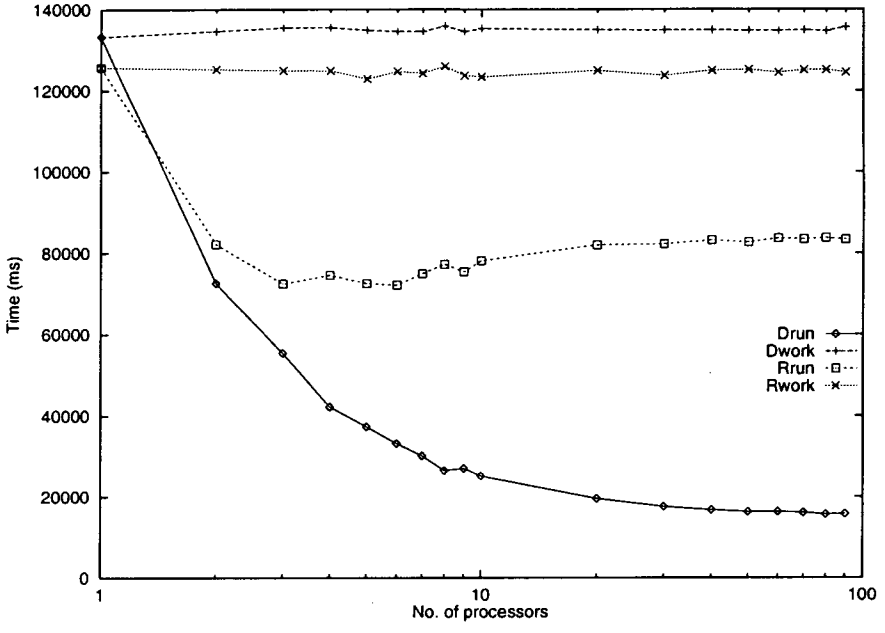


(a) Tsend=20, Tdata=20

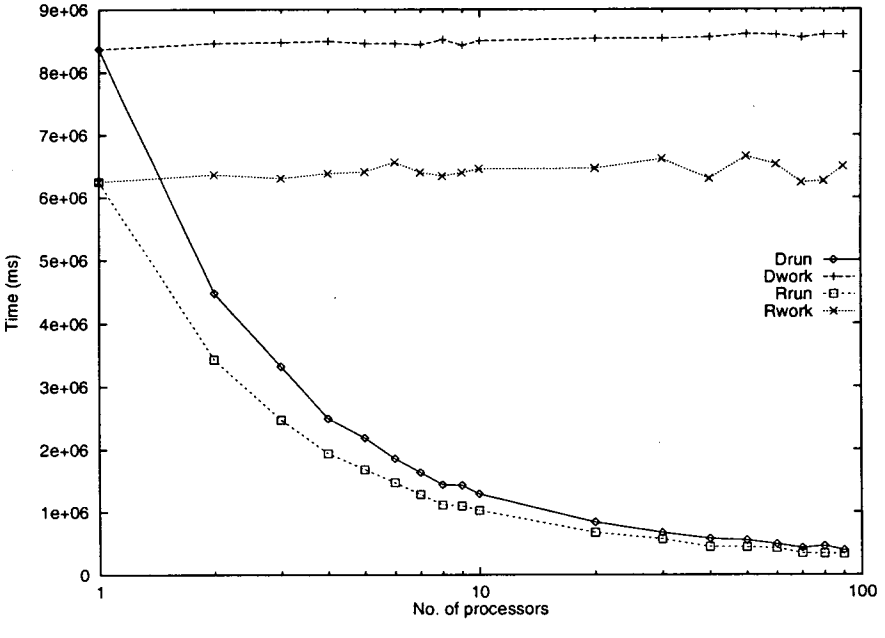


(b) Tsend=20, Tdata=1280

Figure 5.12: Tree 256: Graphs of Completion time and Work performed for differing values of Tsend and Tdata



(a) Tsend=1280, Tdata=20



(b) Tsend=1280, Tdata=1280

Figure 5.13: Tree 256: Graphs of Completion time and Work performed for differing values of Tsend and Tdata (cont.)

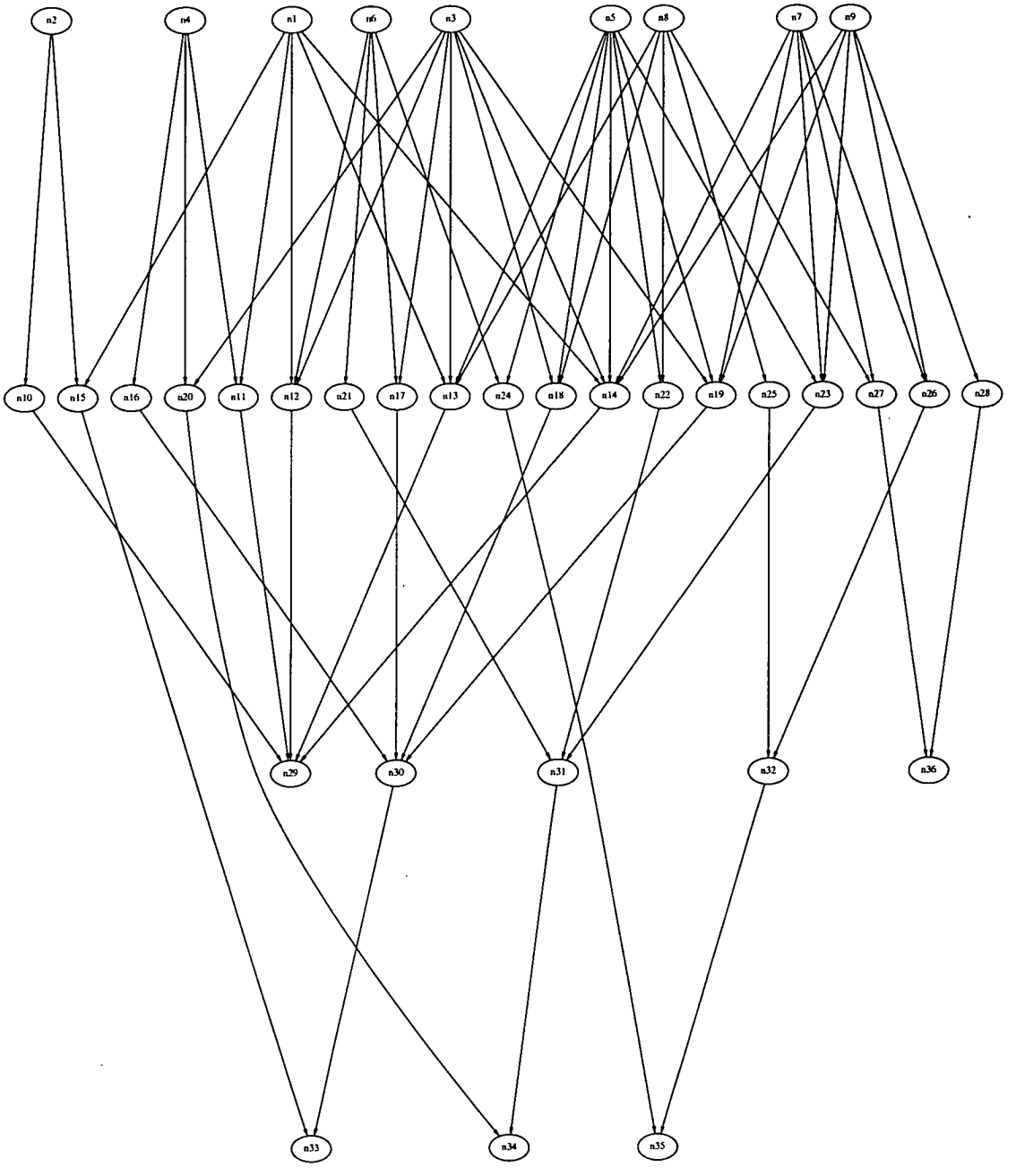


Figure 5.14: Topological layout of the 74LS283 adder

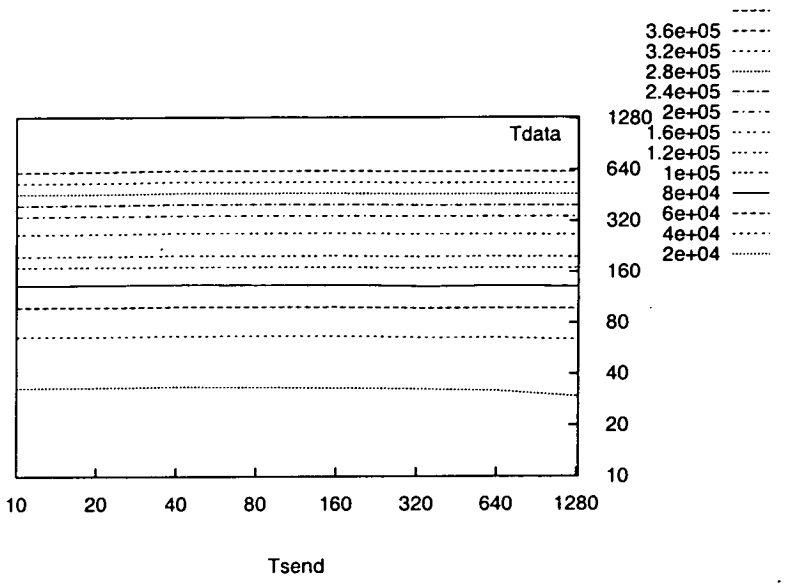


Figure 5.15: Adder: Data-driven runtime as a function of Tsend and Tdata for a 9-processor machine

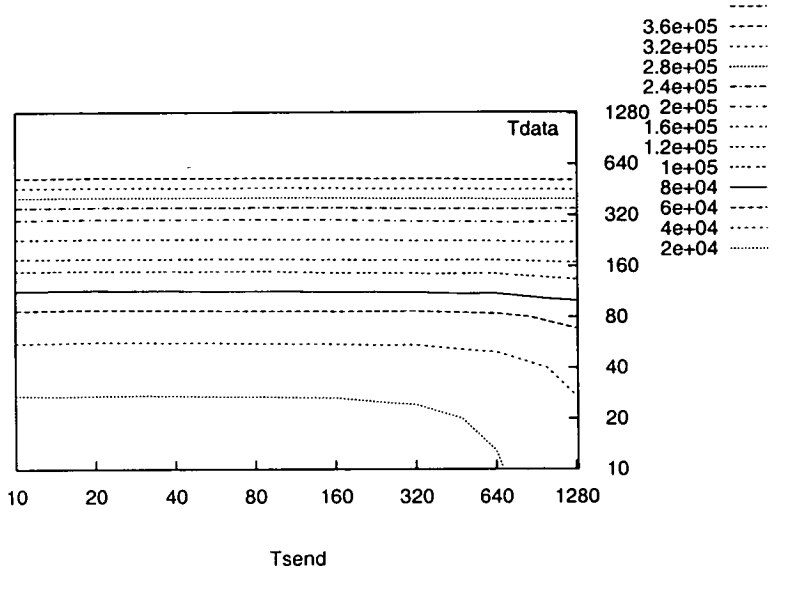
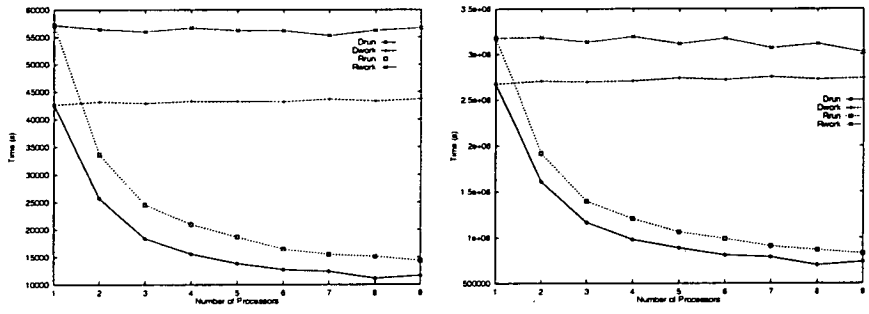
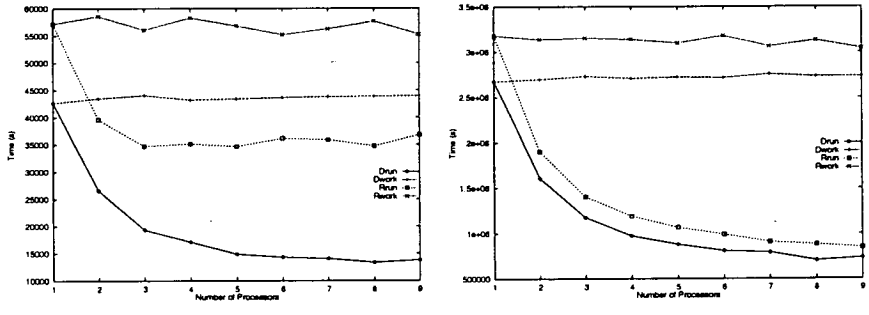


Figure 5.16: Adder: Demand-driven runtime as a function of Tsend and Tdata for a 9-processor machine



(a) Tsend=20, Tdata=20 (b) Tsend=20, Tdata=1280

Figure 5.17: Adder: Graphs of Completion time and Work performed for differing values of Tsend and Tdata



(a) Tsend=1280, Tdata=20 (b) Tsend=1280, Tdata=1280

Figure 5.18: Adder: Graphs of Completion time and Work performed for differing values of Tsend and Tdata

5.4.3 The ISCAS85 Circuits

The ISCAS '85 benchmark circuits[12] are ten combinatorial networks which have been used by many researchers as a basis for comparing results in the area of test generation. Although the circuits were not intended as such, they have also often been used as simulation benchmarks.

Only circuit C880 was simulated. It is an ALU and control circuit with 383 gates, 60 input lines and 26 output lines.

The results presented in Figures 5.21 and 5.22 represent a situation in which the data-driven system consistently out-performs the demand-driven system. In all four graphs, the total amount of work performed by the demand-driven system is about twice that of the data-driven system. While it is encouraging that both systems exhibit similar reductions in run time as the number of processors increases, they appear to level off and it is doubtful if the run time of the demand-driven system would ever fall below that of the data-driven system.

The reasons for such a dramatic performance difference are shown in Figure 5.23. The first graph (Figure 5.23(a)) shows that there is relatively little time for which the demand-driven system is inactive and thus little advantage to be gained from the short-circuit evaluation strategy. This, in itself, would not explain the poor demand-driven performance. The second graph (Figure 5.23(b)) gives a clearer view of the system. The nodes have been ordered so that the maximum number of data messages sent in the demand-driven mode is always increasing (this was done to make the graph clearer). There are two areas of immediate interest: the first is on the left of the graph where the demand-driven system sends no messages. This is caused by the non-evaluation of an entire sub-section of the graph. Therefore, any evaluation performed by the data-driven system is unecessary and not required to determine the end result. The other area of interest, and the one which ensures that the demand-driven system performs poorly, is to the right of the graph. This area shows a very large number of messages being sent down some arcs. Bearing in mind the graph in Figure 5.23(a), which showed that no demand-driven node was required to produce output for the entire simulation run, it is obvious that the average data message size in the demand-driven system

was much smaller than that in the data-driven case. This may well be because of greater interval fragmentation caused by the interaction of both demand and data messages on the state space of the node over time.

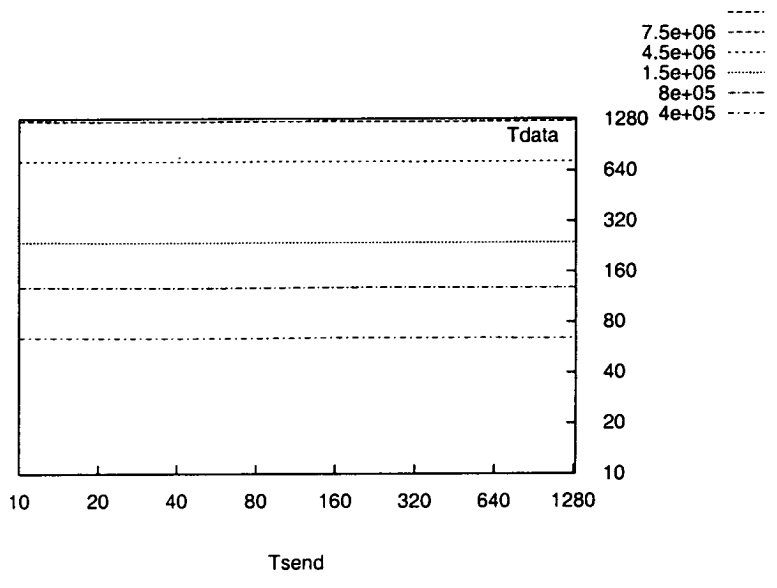


Figure 5.19: C880: Data-driven runtime as a function of Tsend and Tdata for a 9-processor machine

5.4.4 Linear Shift Register

This benchmark was proposed by Greer[37] as a quick and simple circuit which could be constructed easily and scaled to stress the simulating system. The benchmark is constructed by connecting a number of "base units" in series and then putting a feedback loop which, when gated with the input, feeds the first unit. The simplest base unit is a D-type flip-flop.

Shown in Figure 5.24 are N series-connected edge triggered D-type flip-flop units. As shown, these flip-flop units are connected as a shift register with a feedback from unit M. When the value of M is correctly chosen relative to N, and when the output of the last unit is connected to the input of the first, a linear feedback shift register (LFSR) is formed. By repeatedly clocking such a

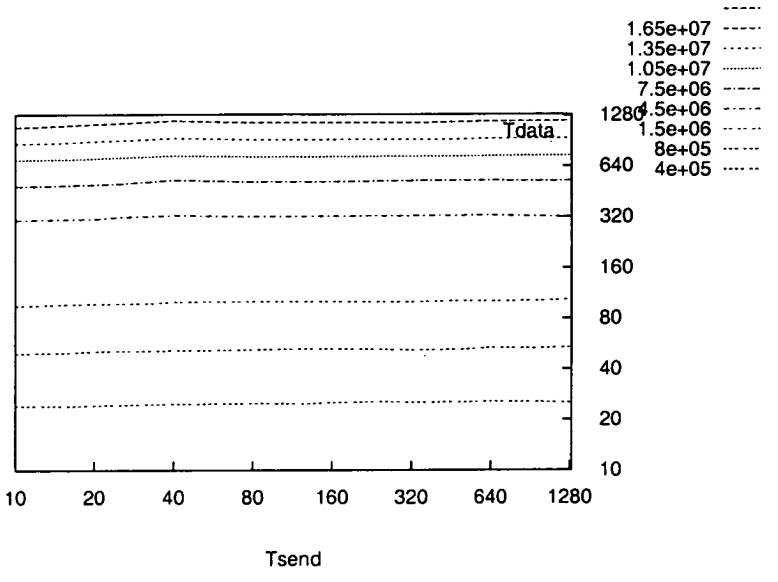


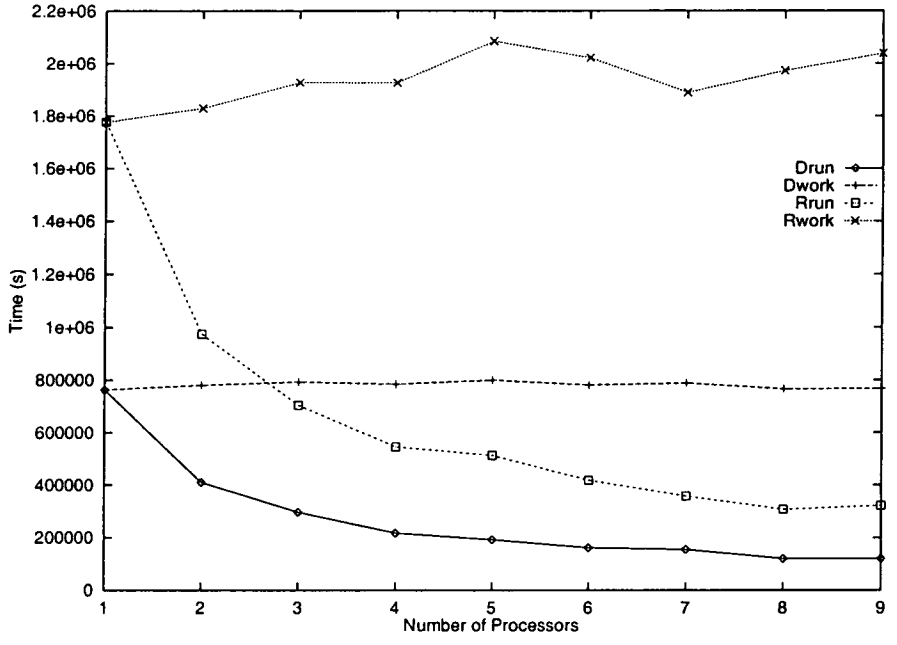
Figure 5.20: C880: Demand-driven runtime as a function of Tsend and Tdata for a 9-processor machine

configuration, following the application of a reset signal, $2^N - 1$ different N-bit words will appear at the outputs of the N units. Additional clock inputs will cause the sequence to repeat.

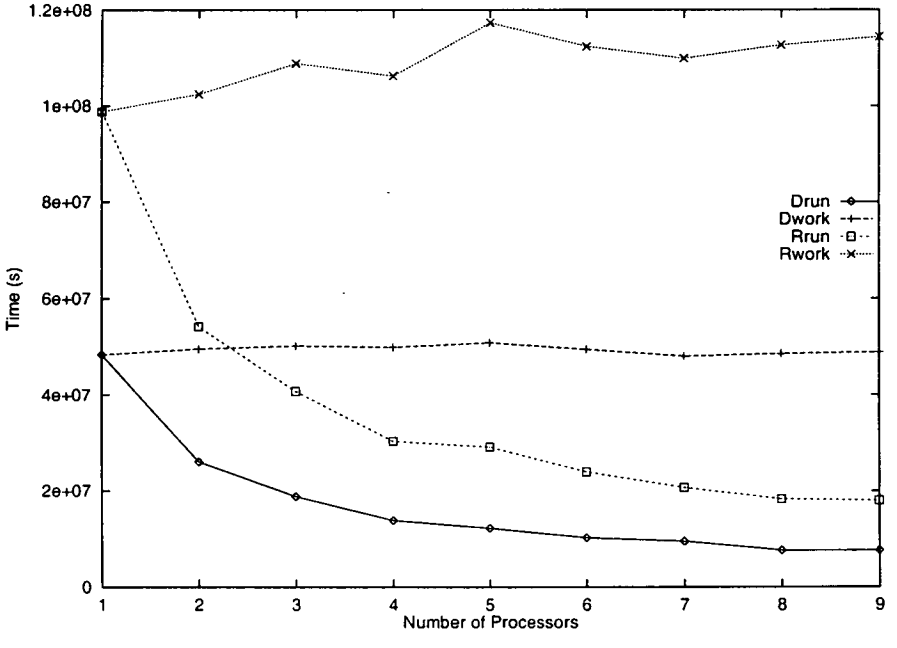
Selected values of N are listed in Table 5.4 along with the location of the feedback unit M. For values of M corresponding to other lengths, see Peterson and Weldon[70].

Figure 5.25 also illustrates that separate LFSR units can be connected in series. Each LFSR has a clock and a data input and a single output. When connected in series, all clock inputs share the same signal while the output from one unit is connected to the data input of the next unit in turn. When this is done, each unit will operate as a separate LFSR. Thus the basic units can be connected in series without limit and, by doing so in hierarchical steps, can create large ciruits with little effort.

The circuit thus created is a pathological example of the effect of feedback on discrete-event simulation. In both the data- and demand-driven cases the

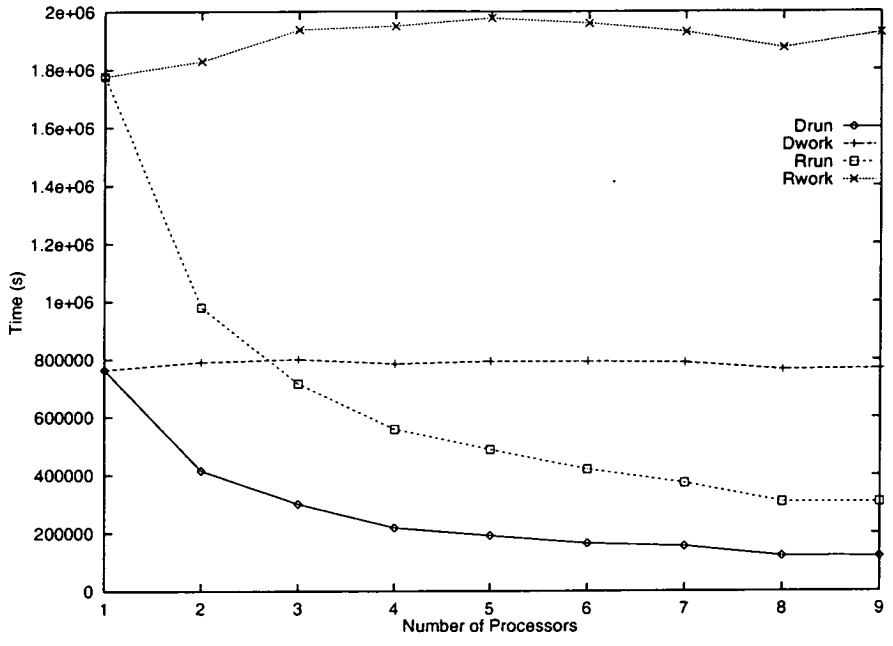


(a) Tsend=20, Tdata=20

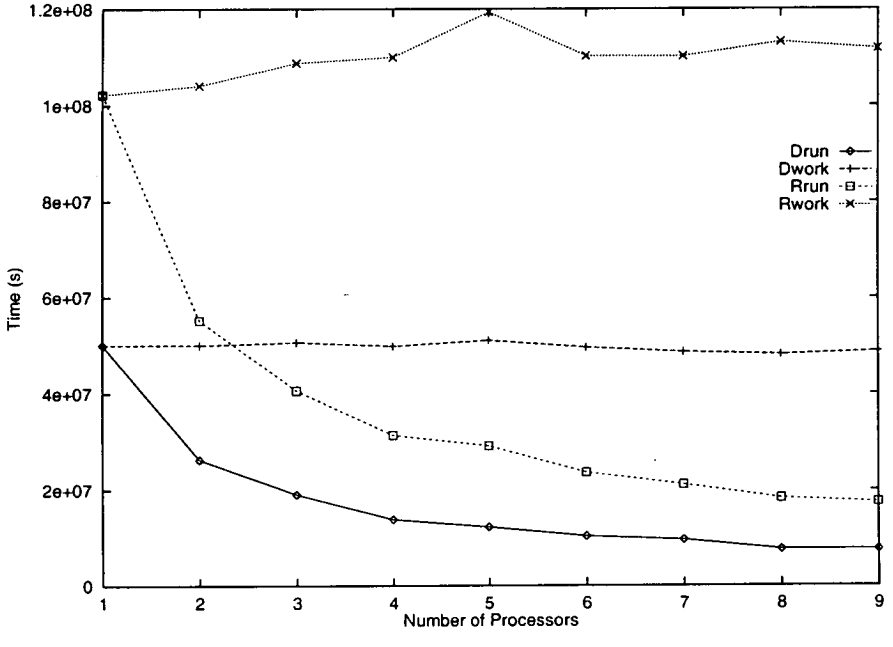


(b) Tsend=20, Tdata=1280

Figure 5.21: C880: Graphs of Completion time and Work performed for differing values of Tsend and Tdata

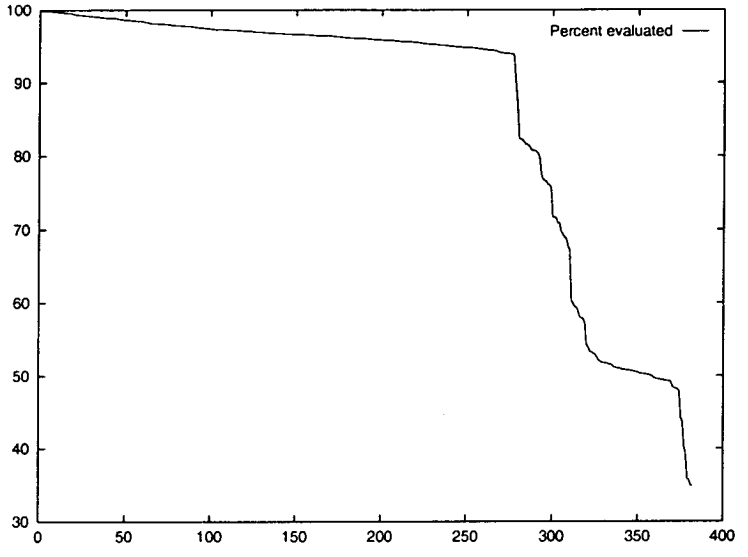


(a) Tsend=1280, Tdata=20

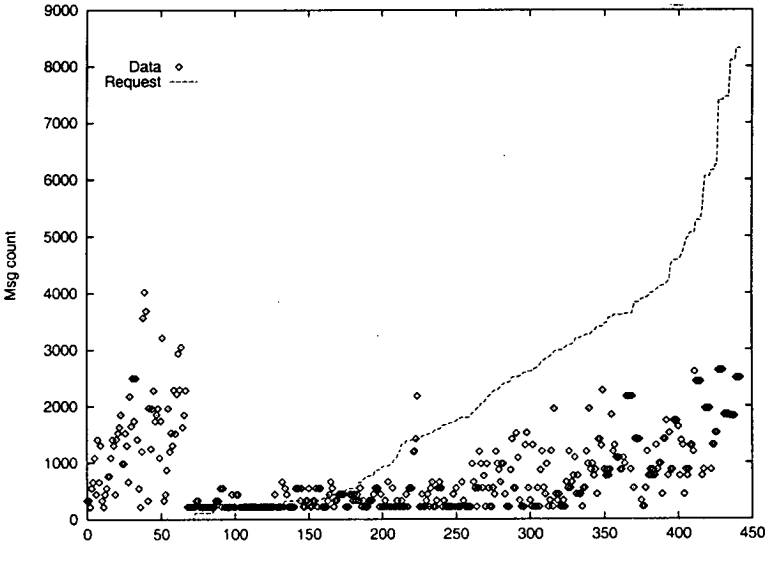


(b) Tsend=1280, Tdata=1280

Figure 5.22: C880: Graphs of Completion time and Work performed for differing values of Tsend and Tdata



(a) Percentage of total time for which a node was active. *Nodes have been sorted to emphasize the unused time.*



(b) Maximum number of messages issued by a node. *Nodes have been sorted in order of increasing demand-driven message count.*

Figure 5.23:

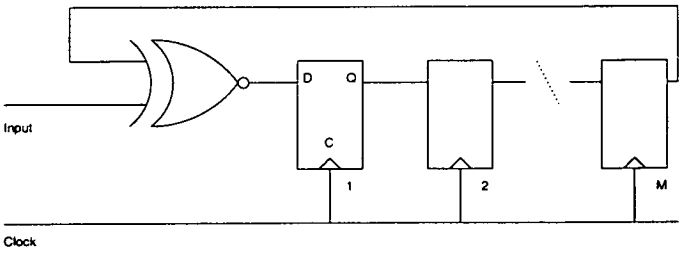


Figure 5.24: LFSR Base Unit

Total Units (N)	Feedback Unit (M)
2	1
3	2
4	3
5	3
10	7
15	14
20	17
25	22

Table 5.4: Tap points to obtain $2^N - 1$ vectors

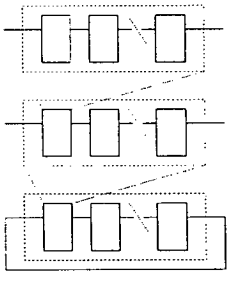


Figure 5.25: Hierarchical Composition of Benchmark

performance is poor. The effect that this has is to divide, and re-divide, the time intervals until the system reaches a state of unit time intervals. In the case of the demand-driven system, as well as the interval fragmentation to be considered, there is also the fact that the base logic elements maintain internal state. Their current state is some function of their previous state and the input applied. This means that, without some reset signal being applied, to determine their current state, we need to evaluate all previous states back to the start of the simulation time. Once such demands for data have been sent, we are in a data-driven mode without any of the short-circuiting, or sub-circuit evaluation, that demand-driven system rely upon for performance.

5.4.5 Causes of Fragmentation

In ELSA, whenever a tuple arrives an output is generated for the fully defined interval. The end time of the output tuple is the minimum of the end times of the input tuples. The net result is that, potentially, the output stream will be more fragmented than the input streams. The amount of fragmentation will depend upon how misaligned the input streams are and the average size of the input intervals.

If the intervals are misaligned by only a small amount then the output stream will consist of tuples representing the misalignment and tuples representing the larger, common, data areas. Should the average interval of one of the input streams be small then the output stream will consists of intervals which are no larger than those of the small interval input stream.

5.4.6 Example of fragmentation

Table 5.5 shows the number of intervals of a given size which occurred in the demand-driven simulation of a 16-element LSR while Table 5.6 shows the number which needed to occur to carry the data. The figures were obtained from an analysis of the data messages sent during the course of the simulation. If two contiguous tuples carried the same state value, then they were combined into a single tuple. This process was repeated until every tuple carried a different state

Size of interval	Count
1	699606
2	45859
3	30809
4	11862
5	2971
6	1786
7	431
8	93
9	146
10	760

Table 5.5: Actual Interval Counts

Size of interval	Count
1	7433
2	5911
3	2856
4	514
5	542
6	795
7	16
8	8
9	7
10	455

Table 5.6: Minimum Interval Counts

to the immediately preceding and succeeding tuple.

The disparity between the figures shows that significantly more messages are actually sent than would be needed if there was no tuple fragmentation.

The weighted mean values for interval size are 13.65 time units and 86.59 time units respectively. The weighted mean is the mean of the product of interval size and count. Given that the input waveform has an average interval of 100 time units and the average gate delay is 3 time units the actual results show a significant degradation of the interval.

Further analysis revealed that the median interval size for the actual results is 1. That is to say that half the messages in the system represent intervals of one time unit. The minimum results have a median of 85. This emphasises the larger interval of the minimum results.

5.5 Conclusions

The general conclusion from the results gathered is that it is possible for the demand-driven system to out-perform the data-driven system. There is a performance gain in that it is possible for the demand-driven system to finish the task sooner than the data-driven system. There is also the gain in resource utilisation in that less of the machine's processor time needs to be used in the demand-driven system. While this result is less important in the area of dedicated single-user parallel machines, in the increasingly common scenario where a parallel resource is shared (such as a network of workstations) a lower resource

requirement may permit more than one simulation run to be performed at the same time (replication-parallelism).

What is, perhaps, more interesting is why the demand-driven system sometimes fails to out-perform the data-driven system. It would appear that the performance of the demand-driven system is very dependent on the structure of the system being evaluated. Even when the demand-driven system performs poorly, many, if not all, nodes are not evaluated for all time. In some cases, the nodes were only evaluated for 50% or less of the simulation time. Also, it is possble to observe that some nodes in a demand-driven system seem to be issuing a far greater number of messages than they do when evaluating under a data-driven strategy.

Coupling the two observations, we can see that some nodes in a demand-driven system are issuing a large number of tuples for very small intervals. As each tuple must be handled, which causes further fragmentation of the tuple stream, the resultant increase in run time is the inevitable result. An improved ordering of nodes for evaluation may help to improve this situation (the current method is only applicable to tree structures and assumes that the order in which nodes are evaluated does not affect the evaluation time of any other node.) Also, various tuple recombination strategies may help to quench the explosion in the numbers of small tuples.

5.6 Chapter summary

In this chapter, a test-bed system was described which was used as a platform for the simulation of both data- and demand-driven simulation systems. A number of logic circuits were simulated on this test-bed and their performance characteristics measured. As data-driven, conservative systems cannot support dynamic topologies, the systems simulated had to be static. Their dynamic nature was examined and we focussed on the effect on parallelism and performance as the available computing resource increased.

The pathological example of a linear feedback shift-register was presented which, as a result of the nested feedback loops, caused both the data- and demand-

driven system to perform badly as a result of the large number of small tuples that both systems sent. The interval size for the data messages was fragmented until both system were, in effect, working with unit time intervals. The demand-driven system was required to request the state of all the elements for all time which then left it operating in a data-driven mode.

Chapter 6

Summary and Conclusions

In this chapter we will summarise our work, discuss our conclusions and give some directions for further work in the area of demand-driven systems.

6.1 Summary of thesis

In Chapter 1, we surveyed the state of the art in distributed simulation and covered the characteristics of the two main systems (conservative – Chandy-Misra-Bryant and optimistic – Time Warp). We noted that neither of these systems has all of the attributes of the desired simulation system.

Chapter 2 looked at the background to the issues and we stepped back from simulation and studied the more generic problems of the production and synchronisation of data in a distributed system. We saw that, by making the receiver responsible for requesting the data rather than have it wait passively, we could obtain a dynamic topology for inter-element connections.

We also clarified the difference between request-driven and demand-driven systems and looked at related work at the micro, compiler and language levels. Related work in simulation, using either demand-driven or request-driven systems, was also discussed.

Definitions of speedup and efficiency, that are widely used throughout the distributed systems field as quality measures, were discussed and we proposed a new measure, opportunity cost, to give an indication of how much of the systems resources are withheld from other potential users.

The chapter closed by looking at binary decision diagrams. This method of

expressing a function can make explicit any non-strictness in that function and as such, is well placed to be utilised in a demand-driven system.

The benefits that are inherent in a demand-driven system were outlined in Chapter 3. It also addressed the more obvious shortcomings of the method and looked at ways to mitigate their effect. Threshold functions were introduced as a tool to address the question of the strictness of a function. The work of Dunne and Leng[31] was introduced as an approach to minimising the work required to evaluate a function. This approach was then expanded to consider the evaluation of the function on a parallel machine through the use of input and output modes.

Chapter 4 described three simulation systems: ELSA [5], a data-driven interval based system, CMB[19], a data-driven event based system and our proposed demand-driven interval based system. Analytical models for the upper bound of the number of messages needed and the processing resource consumed were derived and some suggestions on how to make the upper bound more accurate were made.

The chapter continued with a discussion on the effect of the model's inability to handle non-independent streams effectively. Random binary trees were generated and, as they exhibit independent data streams, were used to predict the expected performance of ELSA and demand-driven system. The results of the models were presented in a number of graphs which show the effect of varying granularity, frequency of events and strictness on both the communications bandwidth required and the processing resources consumed.

Experimental results were presented in Chapter 5. A test-bed system was described which was used as a platform for the simulation of both data- and demand-driven simulation systems. A number of logic circuits were simulated on this test-bed and their performance characteristics measured. As data-driven, conservative systems cannot support dynamic topologies, the systems simulated had to be static. Their dynamic nature was examined and we focussed on the effect on parallelism and performance as the available computing resource increased.

The pathological example of a linear feedback shift-register was presented

which, as a result of the nested feedback loops, caused both the data- and demand-driven systems to perform badly. The interval size for the data messages was fragmented until both systems were, in effect, working with unit time intervals. The demand-driven system was required to request the state of all the elements for all time which then left it operating in a data-driven mode.

6.2 Further work

Throughout the investigation into demand-driven discrete event simulation, a number of questions remained unanswered.

6.2.1 The function/cache dichotomy

The model of demand-driven evaluation which we have used throughout the thesis is one where the node is foremost and the cache of previously computed results is an adjunct to it. That is to say that the node receives and processes the request, either by consulting the cache or by sparking a new computation. This "function centric" view of the system, while simple to implement, is limiting. An alternative approach would be to reverse the view, to treat the system as "smart memory", by putting the cache first as the main recipient of requests and make it responsible for sparking new computation.

The potential advantage of this approach is that, while the memory (cache) would reside in fixed locations, the nodes responsible for calculating the function could be spread throughout the system. This opens up the possibility of relatively fine grain load balancing as each new functional evaluation could be started on the "best" available resource.

6.2.2 Hierarchical evaluation

A simulation model is created from an abstraction of the features of the physical system under investigation. The features chosen for the simulation model may well be some way removed from the physical implementation of those features, e.g. a wire in a digital circuit will have a number of physical attributes (potential, current, capacitance etc.). These will, in some models, be simplified to a single

logical value of true or false. While it is common for the level of abstraction to be uniform across the entire system, and it is less common to have different, but fixed, levels of abstraction within one simulation, standard data-driven methods do not permit the system to alter the level of abstraction dynamically in response to internal conditions. This might be triggered by a condition arising which cannot be modelled successfully at the level of abstraction chosen. A lower level of abstraction would then need to be used. However, to simulate the entire system at the lower level in case such a situation occurred may be impractical due to time or processor constraints.

Demand-driven simulation may be able to address the issue of dynamically altering the level of abstraction in response to local conditions. Should a condition arise that needs to be resolved at a lower level, then the node could request that information at the appropriate level. Further work would be needed to assess the need for such an adaptive system as well as the overhead involved.

6.2.3 Managing load in a peer-to-peer network

Moving away from simulation, an interesting piece of further work would to investigate the extent to which the techniques of demand-driven evaluation can be applied to the emerging peer-to-peer networks.

Assuming a network of web application providers, a user could submit some task to the network. The local node in the network may well be able to perform the work itself or it may choose to send a sub-task to another node.

6.3 Conclusion

Whether the benefits of demand-driven simulation can be exploited depends on the specific situation to which it is applied.

The biggest performance gain in demand-driven simulation comes from non-strict nodes. These nodes give rise to the possibility of not requiring to evaluate large sections of the underlying system graph and thus reduce the amount of work which needs to be performed. This reduction in required evaluation leads, quite directly, through to reduced communication and processor loads. These reduced

loads can often more than compensate for the higher loads imposed by a demand-driven system compared with a data driven system. There is, however, an open question about how strict simulation functions actually are, in practice.

Demand-driven systems also perform well in situations where nodes have a low *active*, output connectivity. Active connectivity relates to links which are actually used rather than being logical connections. This reduces the chances of two nodes requesting similar (but slightly different) time intervals and thus causing increased fragmentation. The more nodes there are requiring the output from another node, the greater the chance of fragmentation. The pathological case of the low output connectivity would be the tree. It is not surprising, therefore, that the tree structure using non-strict nodes shown in Section5.4.1 performed well.

The most striking problem which arises in demand-driven simulation is the potential for an explosion in the number of tuples used to transmit the data between nodes. This problem is the mirror of one which affects the data driven system. The demand-driven system has both to contend with its calendar being fragmented both by request messages and by data messages. If more than one node requests data over two specific (and overlapping) intervals then both requests will be fractured resulting, in all probability, in an increase in the number of tuples sent to *each* requester. This increase can easily swamp any gain achieved by using non-strict nodes.

Determining whether two tuples can be recombined is a relatively trivial task (as long as the states carried can be compared for equality). The only difficulty is to ensure that no deadlock conditions are added to the system by the recombination. The safest place to perform the combination would be in the cache as it does not reply on any other data for its functioning and as such, the worst overhead would be a delay. Tuples can be recombined at other locations as well if suitable buffers are created.

A related issue which requires special handling to operate efficiently is feedback. As in data driven systems, feedback can cause an explosion in the amount of work required to be performed. In the demand-driven case we have the situ-

ation that a node is dependent on its own earlier results to process the current request. This can continue until the node reaches some base case, which may be the initialisation state at the start of simulation time. The node would roll further and further back in time issuing requests and then roll forward again acting as if it was a data driven node. Each request would, in the simplest case, fragment the interval further. The extreme case would have a node stepping forward in time by the delay through the loop even if the data state did not change. This can be handled more efficiently in both data and demand-driven systems.

While demand-driven systems can handle variable (but bounded) delays, it becomes increasingly less efficient at doing so as the bounds on the interval increase. In the extreme example of a node holding a state until some other node sends a signal (quite common in handshaking protocols) then the node holding the signal does not know when the other node is going to send its signal and has, therefore, to hunt back in time until it finds one. Each request causes an overhead as well as increasing the fragmentation. In general, such protocols will hunt back to the start of simulation time and then advance in the normal data driven manner.

While the potential for improved performance depends on the situation to which demand-driven evaluation is applied, there are other gains to be made from using a demand-driven system. The most notable one is partial activation. The need for this feature may be sufficient to make any potential overhead worthwhile carrying.

Partial activation allows the simulationist to place a probe in the system and cause only those nodes whose results are needed to generate a result at the probe to activate. This has the potential to render large parts of the system inactive and thus save on processing power. A similar effect could be achieved in a data driven system by a pre-processing step which could determine all those nodes whose results *might* be needed. The demand-driven system obviates this step as determining the necessary nodes is a function of the basic simulation step.

A side effect of partial evaluation is that the necessary input signals to subcircuits are automatically generated (assuming that suitable signals are available

to the primary inputs). This removes the need (which would exist within a data driven system) of creating accurate sub-circuit input signals.

Demand-driven discrete event simulation, using time intervals, is able to provide a platform with dynamic communication between the nodes, local control of processing, efficiently uses processor power, and is conservative. If the structure being simulated is free from deadlock, then the simulation will be also. The client-server approach means that the evaluation is easy to distribute over available processors. The use of a calendar and time intervals means that the system is able to automatically identify, and exploit, both structural and temporal parallelism in the underlynig system.

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