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# Digital Mixing Consoles: Parallel Architectures and Taskforce Scheduling Strategies 

Ken N Linton, BSc

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Submitted in partial fulfilment of the degree of Doctor of Philosophy


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## Abstract

This thesis is concerned specifically with the implementation of large-scale professional DMCs. The design of such multi-DSP audio products is extremely challenging: one cannot simply lash together $n$ DSPs and obtain $n$-times the performance of a sole device. M-P models developed here show that topology and IPC mechanisms have critical design implications.

Alternative processor technologies are investigated with respect to the requirements of DMC architectures. An extensive analysis of M-P topologies is undertaken using the metrics provided by the $T P G$ tool. Novel methods supporting DSP message-passing connectivity lead to the development of a hybrid audio M-P (HYMIPS) employing these techniques.

A DMC model demonstrates the impact of task allocation on ASP M-P architectures. Five application-specific heuristics and four static-labelling schemes are developed for scheduling console taskforces on M-Ps. An integrated research framework and $D C S$ engine enable scheduling strategies to be analysed with regard to the DMC problem domain.

Three scheduling algorithms - CPM, DYN and AST - and three IPC mechanisms -FWE, NSL and NML - are investigated. Dynamic-labelling strategies and mix-bus granularity issues are further studied in detail. To summarise, this thesis elucidates those topologies, construction techniques and scheduling algorithms appropriate to professional DMC systems.

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## Chapter 1

## Introduction

Trends in audio engineering and VLSI suggest the feasibility of an integrated all-digital music production chain. Clearly, a digital mixing console (DMC) is required to complete this chain.

### 1.1 Background

There has been a requirement to mix musical sound sources from the very early days of recording. Even in the acoustic era vocalists were provided with their own recording 'hom'.

### 1.1.1 Historical Overview

The advent of multi-track tape machines and audio processing 'effects' in the early 1960 s marked a significant turning point in recording and production techniques. ${ }^{1}$ Recording to 'multi-track' meant that there was no longer a need to take irrevocable creative decisions until mix-down. Today, 24 -track analogue and 32 -track digital machines are commonplace with machine transports and MIDI sequencers synchronised via SMPTE timecode.

### 1.1.2 The Recording Studio

Close-miking and multi-track recording techniques are now an integral part of the creative process. To this end, the professional recording studio is divided into two distinct two areas.

### 1.1.2.1 The Sound Room

The sound room is the actual artist performance area and is normally constructed to provide both acoustically 'live' and 'dead' ends: the distinction being the amount of sound absorption in each. Generally, the only technical equipment found in the sound room, apart from electronic instruments and amplification, is monitoring headphones and microphones.

[^0]
### 1.1.2.2 The Control Room

The control room is the nerve centre of the recording studio, where all recording information is routed to be processed and enhanced towards a marketable end product. The control room also houses most of the technical equipment available in the recording studio including the mixing console, multi-track tape machine, outboard effects and monitoring loudspeakers.

### 1.1.3 Production Techniques

With the consumer base increasingly aware of the subtleties of sound balance, studio production techniques are becoming more sophisticated as technical requirements become ever-more stringent. Computer-assisted mixing can be used to manipulate controls set up on previous passes and simultaneously record new control movements. Engineers need only concentrate on a few controls at a time in order to optimise the mix-down process.

### 1.2 Digital Mixing Consoles

Console design is limited both by available technology and conceptual limitations. Not withstanding, DMC design must at least maintain the standards set by analogue counterparts.

### 1.2.1 Design Goals

Goals which the industry has set for console manufacturers include perfect audio response, increased numbers of inputs and outputs, extended creative capabilities, and all at an optimised cost/performance ratio [White, 1988]. It is relatively easy to implement advances in any one, or perhaps a few, of these areas but only at the expense of other performance aspects.

Since standard analogue consoles require mechanically linked control sets, the requirements of more I/O channels together with more creative signal processing capability translate directly into larger desks. Beyond a certain point operational problems, particularly audio quality, parallax and channel strip length cannot be solved by traditional means.

### 1.2.2 Audio Performance

The acuity of the human ear is astonishing, detecting tiny amounts of distortion and yet accepting a dynamic range in excess of 110 dB . While most listeners cannot distinguish an
upper frequency limit, a response down to 20 Hz improves reality and ambience [Muraoka, 1978]. Digital ASP provides the dynamic range and frequency response required.

DSP has replaced and enhanced many ASP functions that were previously accomplished by analogue circuitry: However, the professional audio industry has not succumbed to the apparent digital panacea so readily. Concerns about cost, reliability and audio performance have, to date, limited the acceptance of digital console technology.

### 1.2.3 Control Surface Ergonomics

In analogue consoles, controls have to be positioned close to the actual processing circuitry for performance reasons. One control knob is needed for every variable with the result that the control panel is physically large. Remote control is difficult with such construction techniques and the order of processing stages is determined at the time of design.

Programmability eliminates the need for mechanical links between controls and ASP hardware. The programmability inherent in digital ASP promises an unprecedented degree of freedom for the console designer. Control and display types, and layouts can be optimised strictly for ease of operation, without concern for behind-the-panel physical restriction.

### 1.3 Parallel Architectures

Interest in this area can be traced to the construction of the Cosmic Cube at CalTech. Originally intended for work on planetary dynamics, this M-P found much wider application.

### 1.3.1 Multi-Processors

There are three demarcation points in the spectrum of architectures for parallel computation. At one extreme massively-parallel architectures, such as Thinking Machines Connection Machine, use thousands of simple bit-serial elements. At the opposite extreme, machines such as the Cray $X-M P$ and Fujitsu $V P-200$ rely on several customised vector processing units.

M-P architectures lie in between, offering potentially greater performance in applications with a high degree of intrinsic parallelism. Applications include DSP,
computational fluid dynamics and, more recently, financial modelling. Commercial M-Ps increasingly use 'off-the-shelf' work-station RISC processors, such as DEC's $\alpha$ chip.

### 1.3.2 Processing Technology

Two fundamental decisions are, firstly, how powerful each DSP should be, and, secondly, how many processors should be supported. For a system of required performance $P$, and an ideal architecture of $n$ processors, each with an individual processing capability of $p$, the hyperbolic relationship between $n$ and $p$ defines a span of possible architectures satisfying $P$.

System architects may choose a small number of powerful, expensive processors; or many relatively slow, cheap parts. This latter option is made possible by the emergence of programmable DSP devices suitable for professional ASP. The advantages of using low-cost technology is balanced by the degradation in efficiency that inevitably occurs as $n$ increases.

### 1.3.3 IPC Topologies

Many issues arise from the need for parallel architectures: firstly, some form of interprocessor communications (IPC) mechanism is necessary; secondly, access to shared data must be coordinated. Repeated studies have shown that optimising M-P performance requires a judicious combination of computation and communication performance.

A M-P consists of tens of nodes connected in some fixed topology. Ideally, each DSP would be directly connected to all others. Unfortunately, packaging constraints and hardware costs limit node degree. While many interconnection schemes are feasible, topology metrics enable console designers to quantitatively compare alternative architectures.

### 1.3.4 M-Ps and DMCs

A number of M-P architectures have been built in response to an ever-growing need for supporting computationally-intensive applications. Dedicated architectures aim at maximising performance for a particular taskforce. Within the realm of digital mixing, this approach simply corresponds to direct digital implementation of an analogue console design.

On the other hand, general-purpose architectures are designed to provide good performance for a range of processing configurations. While non-real-time editing is
commonplace in post-production situations, flexible real-time audio mixing is essential for the recording studio environment. It is within this context that M-P topologies are analysed.

### 1.4 Task Scheduling Strategies

Regardless of processing technology and IPC topology, task scheduling strategies are necessary to allocate hardware resources to digital console audio processing algorithms.

### 1.4.1 Introduction

With the availability of fast DSPs with sufficient precision to accommodate the word-growth produced by 16-bit audio [Eastty, 1986], the all-digital successor to today's analogue consoles is now realisable. Several issues pertinent to all large parallel computer architectures are relevant here including code partitioning and task scheduling strategies.

Many of the DMC scheduling techniques discussed here are developed from problems in management science and operations research originally concerned with the utilisation of people, equipment and raw materials. If raw materials are equated with ASP tasks, and people and equipment with processors, then the rationale behind these techniques becomes clear.

### 1.4.2 Partitioning

Partitioning of a console functional description occurs naturally at a level of granularity, appropriate to the large grain dataflow (LGDF) paradigm [Gaudiot, 1988]. To efficiently utilise a DSP computation engine bench-marked at several hundred MIPS and maximise throughput, the DMC functionality must be distributed across available system resources.

Analogue implementations are free from this overhead as the logical structure of the console control surface is fixed by the analogue ASP hardware 'beneath'. In order to shed some light on the complexity of this concept, a systematic study has been undertaken of techniques relevant to the mapping of digital ASP tasks onto multiple DSP configurations.

### 1.4.3 Complexity

Since its introduction in the early 1970's, a wide variety of problems from computer science and operations research are now known to be $N P$-complete. Such problems are now so
pervasive that it is important for anyone concerned with the computational aspects of ASP task scheduling algorithms to be familiar with the meaning and implications of this concept.

Because of the intractability of the task assignment problem, research effort has focused on finding heuristic algorithms that produce sub-optimal assignments and efficient optimal algorithms for restricted cases. Here, these approaches are extended within the context of scheduling mixing console taskforces on large-scale multi-DSP architectures.

### 1.4.4 Opportunities

Such console architectures are inherently software-defined, designed from a pre-defined library of audio DSP kernels. The studio engineer will have the freedom to rebuild the console functionality and re-allocate processing power as appropriate for each session. It is the availability rather than type of processing power which becomes the limiting factor.

Since the audio processing in such a DMC is accomplished through software, the configuration of the desk can be changed at will by running the processing routines for the various functions in a different order. The engineer can configure the DMC to his own requirements by, say, entering symbols on a block diagram displayed on a video screen.

### 1.5 Thesis Overview

Following the discussion of the previous section, this thesis is effectively divided into 3 parts: dealing with digital mixing consoles, parallel architectures and task scheduling strategies.

### 1.5.1 Digital Mixing Consoles

As an introduction to the field of audio engineering, Chapter 2 considers the various forms of ASP algorithm employed in the professional recording studio. An example of 'pop' recording practice and console operation is given together with a review of console terminology and the facilities now considered 'standard' on professional mixing consoles.

This treatment is extended in Chapter 3 to discuss the implications of implementing the mixing function entirely in the digital domain. Automation features are described with the
foundations of control layout and ergonomics developed from the studio engineer's perspective. An indication is given of the form of possible next-generation control surfaces.

### 1.5.2 Parallel Architectures

Chapter 4 discusses the architectural features of programmable DSP devices, serial-bus, ASIC and vector processors with respect to the requirements of digital audio products. Twelve topology metrics are defined in Chapter 5, and an extensive analysis of alternative M-P architectures is undertaken using the Topology Analysis Tool (TPG) developed by the author.

Novel schemes supporting message-passing connectivity for programmable DSPs are compared in Chapter 6. Design decisions taken by the author, leading to the development of a hybrid M-P (HYMIPS) employing these techniques are documented. HSR and HST digital audio peripherals, including schematic diagrams, are also reported in this chapter.

### 1.5.3 Taskforce Scheduling Strategies

A model of DMC scheduling is developed in Chapter 7 and used to explain a variety of speedup characteristics. LPP architectures and IPC mechanisms are then investigated within the same paradigm. A brief survey of deterministic scheduling theory in Chapter 8 leads to the derivation of five application-specific heuristics and four static-labelling schemes.

In Chapter 9, the $D C S$ research framework and predicate hierarchy are described in detail. AST heuristics and CPM labelling schemes are compared using FWE. Dynamic-labelling strategies and mix-bus granularity issues are studied in Chapter 10. Results of this research and further work proposed in the light of these findings are summarised in Chapter 11.

### 1.5.4 Contribution \& Significance

HYMIPS, a hybrid audio M-P targeted at real-time multi-channel ASP, has been developed in accordance with the author's detailed design notes, complete with $H S R$ and HST SP-DIF interfaces. TPG enables direct comparison of 12 M-P topologies within a framework of 13 topology metrics defined by the author in the course of this research. In $D C S$ the author has developed three algorithms tailored to scheduling DMC taskforces on M-P architectures.

Several IPC mechanisms originally investigated from a theoretical perspective are supported including FWE, NSL and NML. CPM, DYN and AST control parameters select from 5 static-labelling schemes, 4 dynamic strategies and 5 admissible heuristics. Together, $T P G$, HYMIPS and DCS contribute directly to the problem domain by advancing fundamental analytical and architectural techniques vital for commercial DMC implementation.

### 1.6 Conclusions

Recent commercial developments have lead to considerable interest in the M-P paradigm for large-scale DMCs. This thesis focuses on salient issues in processing hardware, system architecture and software allocation for such products. The goal is to provide insight into the major design decisions which must be addressed during the development of such a console.

The development of M-P topologies clearly depends on the recent appearance of powerful single-chip DSPs and inexpensive memory. Although M-Ps offer potentially enormous computational power at modest cost, many important questions must be resolved before DMCs can exceed the cost/performance ratio provided by today's analogue consoles.

As yet, there is no expedient solution. Although DSP processors are available with the required level of processing power, audio processing and operating system software, and the IPC to couple it must all be developed by the console manufacturer. This is a massive undertaking, but there is no other way to fully realise the future of audio console design.

### 1.7 References

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## Chapter 2

## Analogue Mixing Consoles

The mixing console - also called a 'desk' or 'board' - is the heart of the recording studio. Once an audio signal enters the console, everything that happens to it before reaching the final medium, including level control, can be described as audio signal processing (ASP).

### 2.1 Audio Signal Processing

The primary ASP techniques are those which modify the frequency spectrum, dynamics, time domain, or spatial location. Figure 2-1 shows how they may be classified.


Figure 2-1: Classification of audio signal processing techniques.

### 2.1.1 Frequency Spectrum

In the infancy of sound recording the goal was to achieve an essentially 'flat' frequency characteristic. Equalisation (EQ) arose as a means of correcting deficiencies in the response.

### 2.1.1.1 Filters

Sharply attenuating frequencies above and below the spectral range of a musical instrument reduces 'leakage'. ${ }^{1}$ For example, a 10 kHz low-pass filter attenuates hiss-type circuit noise without affecting tone quality as much as gradual treble roll-off.

### 2.1.1.2 Parametric Equalisation

The most typical EQ employed in mixing consoles is the parametric equaliser. Parametrics allow continuous adjustment of centre frequency, gain, and tightness (or $Q$ ) of the response curve. As illustrated in Figure 2-2, $Q$ can be adjusted to act on one narrow frequency band or to have a very broad effect. Overlapping bands make this type of equaliser very flexible.


Figure 2-2: Range of control provided by one section of parametric equalisation.

### 2.1.1.3 Graphic Equalisation

A row of slide potentiometers divides the audible spectrum into bands centred at frequencies determined by ISO recommendations - a third-octave unit has 31 separate bands. A large

[^1]number of bands can result in considerable phase response degradation. Rarely found within professional consoles, graphic EQ is often used as outboard processing.

### 2.1.1.4 Psychoacoustic Processing

Psychoacoustic processors increase presence without the side-effects, such as increased noise, inherent in conventional EQ. Units such as the Aphex Aural Exciter add a low-level signal of harmonics not present in the original, generated via distortion with high-pass filtering.

### 2.1.1.5 Studio Applications

EQ is used in the studio for two distinct but overlapping purposes. In the first, EQ is used to compensate for frequency response aberrations, particularly those which result from cardioid microphone placement. On an orchestral session, this kind of manipulation may be quite sufficient. Conversely, on a pop session, the engineer will make an unconscious transition from using EQ in a 'traditional' sense to using it for purely creative purposes.

### 2.1.2 Dynamic Range

Dynamic range control is a vital part of the recording process. Whilst riding a fader is still one of the most effective methods, engineers have come to rely on automated ASP techniques.

### 2.1.2.1 Compression and Limiting

Compressors reduce the gain applied to an input when the signal level exceeds a preset 'threshold'. Dynamic range is reduced while transient peaks are passed unaltered. Musically useful ratio settings range from $1.5: 1$ to $4: 1$. In comparison, limiters apply extreme compression to transient peaks to retain 'musical' events without altering dynamic range. ${ }^{1}$

### 2.1.2.2 Expansion and Noise-Gating

In ASP terms, expanders and noise-gates are simply the inverse of compressors and limiters, A gentle characteristic will gradually increase the attenuation as the levels falls; a harsh ratio, close to $\infty$, as in a noise-gate, will sharply reduce the apparent noise level. Noise gates usually allow access to the side chain to enable the gate to be controlled by a separate 'key' signal.

[^2]
### 2.1.2 3 Studio Applications

Again, applications can be divided into two main areas: control of dynamics to enable 'difficult' material to be recorded, and dynamics control for creative effect. For example, a slow compressor attack time can be used to 'tighten-up' a bass guitar as the deliberate overshoot on initial transients adds punch. During mixdown, a gate is sometimes used on each output of an analogue equipment to reduce tape hiss without removing program material.

### 2.1.3 Time-Domain Processing

Modern music production requires numerous time-domain techniques. Central to these effects is the ability to delay audio signals, and then recombine original and delayed waveforms.

### 2.1.3.1 Basic Delay Effects

If the original signal is delayed by 50 ms to 1 s two distinct sounds are heard. A short delay of around 200 ms results in 'slap echo', as originally produced by tape-loops in the 1950s for rock ' $n$ ' roll records. If the delay time is reduced to around 15 to 35 ms , delays can be used to 'double' a sound, creating an effect known as automatic double tracking (ADT). ${ }^{1}$

### 2.1.3.2 Modulated Delay Effects

If the delay time used in ADT is modulated by a low-frequency oscillator (LFO), slight pitch detunings are produced like those which occur in a chorus of unison voices. At delay times less than 20 ms , the ear is unable to resolve direct and delayed signals. Phase cancellations result in a series of dips in the net frequency response known as flanging (see Figure 2-3).

### 2.1.3.3 Pitch-Shifting

A pitch-shifter is a delay device which can change the pitch of a signal in real-time without changing its duration. In professional units with well-designed algorithms, such as the Eventide Harmonizer, shifts of up to 2 octaves are possible [Bogdanowicz, 1989]. Pitchshifting allows the creation of harmony parts or the correction of out-of-tune notes.

[^3]

Figure 2-3: Example of effect of flanging on flat frequency spectrum.

### 2.1.3.4 Digital Reverberation

To simulate the highly diffuse reverberant tail of natural reverberation (see Figure 2-4), a comb and all-pass filters can be used together with recirculating delays. ${ }^{1}$ Another method is to trace the paths of individual rays emanating from an isotropic source. Unnatural effects, such as non-linear and reverse reverb, are now fully integrated into the studio engineers' repertoire.

### 2.1.3.5 Studio Applications

A wide variety of musical effects can be created by simply delaying an audio signal. For example, several repetitions can be evenly spaced creating an echo rhythm in sync with the original track. ADT imparts a sense of ambience to close-miked instruments not unlike early reflections. Chorusing can be used to split a mono sound into pseudo-stereo. Digital reverberation is employed to give an authentic sense of space to acoustically dry recordings.

### 2.1.4 Spatial Localisation

Classical psychoacoustics predicts that interaural intensity difference (IID) and interaural time delay (ITD) are the primary cues used by the human auditory system [Rossing, 1990].

[^4]


Figure 2-4: Effect of artificial reverberation unit on sound impulse.

### 2.1.4.1 Localisation by Level

The panoramic potentiometer or pan-pot gives constant loudness for a single source between stereo channels, as Figure 2-5. This type of spatial manipulation is more accurately referred to as panned-mono, as sound localisation is determined solely by IID. Pan-pot design is a surprisingly complex area, primarily, because it is necessary to retain mono-compatibility.

### 2.1.4.2 Sum-and-Difference Panning

An extension of level-based panning, this method relies on a sum-and-difference matrix encoder and decoder, as shown in Figure 2-6. This effect is often found in consumer portable stereo systems to generate a 'wide stereo' effect. Although capable of placing a sound beyond the width of the loudspeakers, compatibility with mono is severely limited.


Figure 2-5: Example of typical console pan-pot characteristic.

### 2.1.4.3 Commercial Spatialisation Systems

Spatialisation techniques emulate the directional cues used by the human auditory system, combining level and phase cues together with head-related transfer-functions (HRTF). ${ }^{1}$

### 2.1.4.3.1 QSound

This system combines delay with selective manipulation of the signal in the frequency domain - $6-8 \mathrm{~dB}$ of boost/cut in narrow 11 Hz bands. A total of 8 inputs leads to either 4 placement positions on each side or continuous panning across the soundfield. Sound sources can be placed with high resolution in a $180^{\circ}$ arc in front of the listener [White, 1992].

### 2.1.4.3.2 Roland Sound Space - RSS

RSS provides optimum loudspeaker reproduction via transaural processing. Crosstalk is eliminated by subtracting left-right and right-left crosstalk components from the originals. Left and right signals are then processed using a set of FIR filters and weighting coefficients. In practice RSS works best between 500 Hz and 2 kHz as it can occasionally disturb timbre.

[^5]

Figure 2-6: Outline arrangement for sum-and-difference panning.

### 2.1.4.3.3 Dolby Stereo

Both Dolby Stereo, and the Pro-Logic surround system split the signal into 4 channels: LEFT, RIGHT, CENTRE and SURROUND. The CENTRE channel consists of dialogue only, whereas ambience and special effect information are assigned to the SURROUND channel. It should be noted that SURROUND channel is limited to a bandwidth of 100 Hz to 7 kHz .

### 2.2 Console Terminology

This section describes the elements found in a typical recording console. While every manufacturer's terminology varies to some degree, most include the features described here.

### 2.2.1 Multitrack Recording Techniques

Over the years fundamental differences have developed between multitrack recording techniques used for 'classical' and 'popular' repertoire. As popular music is unquestionably the most demanding of console facilities, a method of 'pop' recording is described below.

### 2.2.1.1 Track-Laying

For the purposes of tracking, instruments are not all normally recorded simultaneously. To allow maximum flexibility in the later stages, the multitrack master is built up a few tracks at a time. ${ }^{1}$ On the first 'take' the basic rhythm tracks including drum kit, bass guitar, rhythm
guitar and guide vocal will be laid down using a minimum of processing. Figure 2-7 shows the complexity of microphone arrangement typical for a studio drum kit.


Figure 2-7: Typical arrangement of microphones around drum-kit (courtesy [Bartlett, 1988]).

### 2.2.1.2 Overdubbing

Successive takes are used to 'overdub' further instrumental parts in sync with those already on tape until the lead vocal is re-recorded. The signal path is the same as that used in tracking, except that microphone signals are mixed with tracks already on tape. In order to avoid microphone 'spillage', it is usual to acoustically isolate performers from one another. Studio 'cue' headphones provide a balanced mix of live and recorded tracks.

1. To enable synchronisation to MIDI equipment, the tape is first striped with SMPTE timecode along its length.

### 2.2.1.3 Mixdown

At some later date the multitrack master is mixed down to stereo. The engineer will add various effects processing and change, often dynamically, the relative balance and position of instruments. As a result, the final sound is often very different to that of the monitor mix heard during previous stages. Input channels are often used as channel processing facilities are usually more elaborate than those provided in the track monitor paths [Bartlett, 1988].

### 2.2.2 Two-Channel Console

Figure 2-8 shows the signal paths in a basic two-channel console. ${ }^{1}$ After the microphone signal is properly pre-amplified it passes through EQ and, perhaps, other dedicated processing sections. A 'fader' - usually a linear ${ }^{2}$ potentiometer - provides level adjustment, while a pan-pot simulates IID in the panned-mono sound-stage of multitrack recording.


Figure 2-8: Signal flow diagram of a typical 2-channel mixing console.

Signal may also be sent to external processing, such as a reverberation unit, by using the 'echo-send' or 'aux-send' control. The 'echo-return' adjusts the overall level of

[^6]reverberation added to the mix as a whole. Note that the signal path illustrated in Figure 2-8, with inputs from tape tracks, is essentially that of a multitrack console during mixdown.

### 2.2.3 Simple Multi-Channel Console

Figure 2-9 shows the signal paths in a typical multi-channel console. The design is similar to the 2 -channel console described above, but with the addition of channel assignment routing.


Figure 2-9: Signal paths in a typical $m$-channel console (during recording). ${ }^{\text {a }}$
a. For clarity, all level controls are shown as linear faders: however, many will be rotary. PPM represents peak programme meter, one type of professional console metering ballistic (see Section 2.2.4 below).

### 2.2.3.1 Signal Routing

Controls that affect a single input are termed a 'module' or 'channel strip'. In conventional designs this is generally a vertical panel in the control surface. A typical desk consists of $n$
such modules, output groups and some form of metering. ${ }^{1}$ Several microphones (say, for a brass section) can be grouped and panned across a number of outputs, to create a 'sub-mix'.

### 2.2.3.2 Solo and Cut

The module 'solo' button allows the engineer to monitor this input only, without affecting other console functions. As the main monitor signal is momentarily muted, soloed inputs are still heard at their proper levels and stereo positions. The 'cut' switch mutes the input by disconnecting the module output from the channel fader, routing and post-fader aux sends.

### 2.2.3.3 Mix Buses

As shown in Figure 2-9, there are also two subsidiary mixing circuits in any console. During recording, the 'monitor' mixer allows the engineer to preview the sound of the final master without affecting signals going onto tape. The 'cue' mixer combines pre-recorded tape tracks and live microphone signals onto the cue mix-bus sent to the musicians' headphones.

### 2.2.4 Metering

Console level meters are of two types: volume unit (VU) and peak programme meters (PPMs). PPMs are almost linear in dB and able to detect momentary transients, whereas VUs give low readings on music programme. Mechanical meter movements have been superseded by neon plasma or LCD columns, with electrically simulated ballistics. Plasma meter systems often display stereo spectrum analysis or channel VCA display during automated mix-down.

### 2.2.5 Free-Grouping

With free-grouping, any channel may be designated as a sub-group. Several channels, $1 \& 2$ in Figure 2-10, are directed to mixbus 1 which feeds the 'group' channel 3. This output, and other inputs such as channel 4, are combined on mixbus 2 with the last channel controlling the group output. This concept is common in 'in-line' multitrack consoles to permit varying programme use. Switching is arranged so that a channel cannot pick up its own group.

[^7]

Figure 2-10: Illustration of the free-grouping principle.

### 2.3 Professional Console Features

Most professional consoles use an I/O type of construction, also termed 'in-line'. The following sections discuss those features found on such professional recording consoles.

### 2.3.1 Input/Output Module

For example, the SSL $G$ Series SL 611G I/O module has two completely independent audio signal paths - CHANNEL and MONITOR. ASP sections may be switched into either path.

### 2.3.1.1 Filters and Equalisation

This section combines high-pass (H-P) and L-P filters, Figure 2-11, with four-band EQ, Figure 2-12. High-frequency (H-F) and L-F sections comprise $12 \mathrm{~dB} /$ octave shelves with variable cut-off frequency and boost/cut. Continuously variable Q ( 0.5 to 3.0 ), gain ( $\pm 15 \mathrm{~dB}$ ) and centre frequency (12:1) are provided in the high-mid-frequency (HMF) \& LMF sections. $\times 3$ and $\div 3$ buttons enable the HMF and LMF sections to work with or against the H-F and L-F sections. The EQ and Filters can be routed separately to the signal paths within the module using SPLIT. This approach allows the Filters to be used in the CHANNEL, which feeds the multitrack while in RECORD, with the Equaliser used on the MONITOR path only.


Figure 2-11: SSL G Series I/O module Filters section (courtesy SSL Ltd.).


Figure 2-12: SSL G Series I/O module EQ section (courtesy SSL Ltd.).

### 2.3.1.2 Dynamics

The dynamics section of an SSL $G$ Series, shown in Figure 2-13, comprises a compressor/ limiter and an expander/gate using the same VCA. Both sections work independently but can
be operational at the same time to provide dynamics control of either signal path. Switching links the side-chain signal to that of the next Dynamics section for stereo operation.


Figure 2-13: SSL G Series I/O module Dynamics section (courtesy SSL Ltd.).

### 2.3.1.3 Cue and Aux Sends

As shown in Figure 2-14, the SSL Cue and Aux send system is based around one stereo and four mono sends per module. Each send can be derived pre- or post-fader (PRE) and from the MONITOR signal path, instead of the CHANNEL signal path, with the SMALL FADER button. The stereo send, normally used for musicians' cue headphones, also has a pan control.

### 2.3.2 Centre Section

Typically, the centre section is designated for machine control, automation management and log-keeping. The centre section of a $G$ Series is fitted with a full-sized QWERTY keyboard, as Figure 2-15, including an in-built TV monitor to display automation and other housekeeping


Figure 2-14: SSL G Series I/O module Cue and Aux Send section (courtesy SSL Ltd.).
information. The SL 651 G Master Facilities Module houses master logic, monitoring and metering control; aux send and return masters; and talkback and listen microphone systems.

### 2.3.3 Master Routing and Signal Flow

Six key points in each I/O module define two separate audio paths: CHANNEL and MONITOR input sections, SMALL and LARGE faders, and two outputs. One path, the CHANNEL, is normally controlled by the LARGE (VCA) fader. Global status buttons determine the default signal paths of all I/O modules, supporting configurations from track laying to final mixdown.

There are four basic desk statuses: RECORD, REPLAY ${ }^{1}$, MIX and RECORD+MIX (overdub). In RECORD or REPLAY, all inputs switch to MIC, whereas in MIX all inputs switch to

[^8]

Figure 2-15: SSL G Series Master Facilities Module and Centre Section (courtesy SSL Ltd.). LINE for final mixdown, allowing SMALL faders to be used for additional inputs. RECORD+MDX reverses the normal MIC/LINE selection for selected channels for overdubbing purposes.

### 2.3.4 VCA Grouping

The LARGE fader sends a control voltage, via the studio computer, to a VCA in the I/O module. As a result, the automation system can automate the level of the LARGE fader, Figure 2-16, usually fed from the CHANNEL signal path. Control of the LARGE fader is assigned to one of the eight VCA group faders using the channel VCA thumbwheel.

VCA grouping allows several channels to be grouped at the control, rather than audio, interface. For example, several independent tracks in a backing vocal mix can be grouped while the individual audio signals remain intact. These channels may be controlled by the same fader, so easing problems of physical reach and simultaneous operation.

### 2.4 Console Automation

According to Jones \& Jubb, "The mix-down process is a search for perfection" [Jones, 1987]. Every recording engineer has encountered the frustration, usually experienced in the early hours of the morning, of searching for 'that mix' achieved 8 hours previously.


Figure 2-16: SSL G Series I/O module Large Fader section (courtesy SSL Ltd.).

### 2.4.1 Automated Mix-Down

Console automation reduces the effort required to perform routine studio operations. Virtually all high-end music production consoles have extensive computer-assisted mixing facilities.

### 2.4.1.1 Comparison

An inherent disadvantage of VCA automation is that the physical fader position bears no direct relation to absolute gain. Despite visual aids, the loss of this direct relationship presents
a serious perceptual difficulty. An alternative technique, employed in moving fader systems such as Neve's Flying Faders and SSL's Ultimation, is to drive the fader itself. This approach recognises that the fader knob is simultaneously a means of control and an indicator. ${ }^{1}$

### 2.4.1.2 Synchronisation

SMPTE time-code is recorded onto one tape track and used to relate mix data to tape position. Computer-assisted mixing manipulates the controls set up on previous passed and simultaneously records any new movements. Such mix automation systems, as exemplified by the Neve $V R$ and SSL $G$ Series desks, enable the engineer to recall previous combinations at will, and use the best of previous mixes together with just a touch of final adjustment.

### 2.4.2 Beyond Fader Automation

In order to achieve complete automation of the professional analogue console, programmable equivalents must be found for both switching and variable functions.

### 2.4.2.1 Switching Functions

FET switching is sufficiently reliable and cost-effective for professional audio use. Using nonlocking buttons controlling FETs, it is possible to build analogue consoles in which all simple switch functions are automated. However, FET switches can inject 'charge transfer' clicks into the signal path, presenting considerable problems when full programmability is sought.

### 2.4.2.2 Variable Functions

In order to automatically reset potentiometers to within $2 \%$ error, it is necessary to find some means of either remotely controlling potentiometers or developing a programmable replacement. To a limited extent, both approaches have been successfully implemented in moving-fader or VCA schemes: both have gained widespread acceptance within the industry.

Many variable functions are achieved by rotating knobs and variable potentiometers. The costs involved in replacing all such standard control sets with VCA or servo-motor

[^9]technology are, however, staggering [Saunders, 1985]. Control range stability in VCAs and repeat-accuracy tolerances in servo-motors restrict the usefulness of either substitute.

### 2.4.3 Total Recall

Memory and automation can today extend further than the faders alone. The Total Recall system, pioneered by SSL, employs an additional track on each potentiometer and an extra pole on each switch, to provide positional information. ${ }^{1}$ As the controls are manual, there is of course no automatic reset, but a colour VDU facilitates manual resetting by the operator.

To go beyond Total Recall, requires automatically resetting over 1,500 potentiometers and possibly double that number of push button switches. While digitally-controlled analogue designs - such as the Euphonix $C S$ - do exist, the desire for greater automation capability poses a virtually insurmountable challenge to analogue mixing technology.

### 2.5 Conclusions

ASP has defined the characteristic recorded sound of each musical era. The 1950s had valvebased compressors and slap echo; the 1960s used 'fuzz' distortion, wah-wah, and flanging; the 1970s popularised delay and psychoacoustic processors; in the 1980s recording techniques were revolutionised by digital reverberation. Today, spatialisation systems are stressed.

Professional mixing consoles contain considerable processing facilities, notably EQ. Several high-end products also offer control of dynamics and some have offered a degree of time-domain control. Basic control of spatial parameters is provided by the humble pan-pot. At the present time more comprehensive processing requires outboard equipment.

DSP has added to the repertoire of sounds available in the recording studio, while making effects that were originally difficult to obtain now relatively easy. Digital ASP has taken full advantage of decades of evolution in the computer industry and is fortunate in benefiting from contributions from such otherwise unrelated fields as seismology and radar.

Console operation would be simple - and totally inflexible - if the signal paths were fixed. As it is, there are many permutations which give professional consoles the flexibility

[^10]accepted as an essential part of audio mixing. These permutations allow the engineer to configure the desk to conform to any task presented during a recording session.

The industry demands a greater numbers of inputs and outputs with more creative signal processing. Size, number of controls, and component density are constant design problems as analogue consoles have kept pace with operational requirements. It is unlikely that these problems can be resolved while maintaining standard analogue performance levels.

Rather than developments in ASP as previous decades, the author predicts that advances in professional console technology will define recorded sound in the coming epoch. While some digital consoles exist in the current marketplace, none provide the degree of functionality beyond programmable analogue originally anticipated by the industry.

This issue depends very much on the design of the control surface itself, and the power and reliability of control and automation software. These factors are especially critical in the evaluation of any digital console. Consequently, it is appropriate to investigate the issues of digital implementation as they affect the console designer and, ultimately, the DMC operator.

### 2.6 References

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## Chapter 3

## Console Digitalisation

In any DMC, three sub-systems can be identified as shown in Figure 3-1. The control/display surface sends commands to - and displays data from - the control/display processing, which determines the control parameters to be conveyed to the audio signal processing.


Figure 3-1: Block schematic representation of generic DMC.

### 3.1 Discrete-Time Signal Processing

In order to manipulate an audio signal in the digital domain, it must be first broken into evenly spaced time elements - sampling - and then expressed in binary form - quantisation.

### 3.1.1 Sampling

In Figure 3-2(a), a high sampling rate is intuitively adequate whereas in Figure 3-2(b) the output waveform (shaded) is at a new, lower, frequency. To explain this phenomenon fully the sampled signal must be studied in the frequency domain.


Figure 3-2: Illustration of the aliasing phenomenon in the time domain.

### 3.1.1.1 Periodic Sampling

Since the sampling impulses, $f_{s}$, are vanishingly short in duration, their spectrum is infinite as shown in Figure 3-3(a). The input amplitude-modulates these impulses, producing upper and lower side-bands as Figure 3-3(b). In Figure 3-3(c), $f_{s}$ is too low for the input bandwidth, and there is a region of overlap (shaded) between the base-band and mirror image about $f_{s}$.

### 3.1.1.2 Aliasing

In Figure 3-3(d), reconstruction can no longer separate the resulting ambiguities. This phenomenon, termed aliasing, is similar to stroboscopic effects observed in rotating objects in motion pictures. Aliasing can be avoided only if $f_{s}$ is greater than twice the original signal bandwidth, $f_{b}$. Typically the input is band-limited with a brick-wall low-pass filter.

### 3.1.1.3 Choice of Sampling Rate

The slope of available L-P filters forces designers to raise $f_{s}$ above the theoretical minimum. In consumer products, rates close to twice 20 kHz are used. For vari-speed operation a higher rate is required since the first image frequency could pass the reconstruction filter as off-tape sampling rate falls. 48 kHz has been adopted for professional use as it has a simple relationship (3/2) to 32 kHz FM landlines and is far enough above 40 kHz for vari-speed.

### 3.1.2 Quantisation

Uniform quantisation divides the continuous range up into equal intervals of width $Q$. Since $n$ bits may represent at most $2^{n}$ different values, the closest $n$-bit value must be determined.
(a)


Figure 3-3: Illustration of the sampling process in the frequency domain.

### 3.1.2.1 Quantisation Levels

Whatever the exact analogue input value, it will be represented by the number at the centre of the interval in which it falls. The true continuous analogue voltage differs from its binary representation by an error which cannot exceed $\pm 0.5 Q$, as detailed in Figure 3-4. This additive random noise is termed quantisation noise, the digital equivalent of analogue thermal noise.

### 3.1.2.2 Quantisation Noise

If the probability density function of the quantisation noise ${ }^{1}$ is assumed to be uniform, a simple connection can be made between $n$ bits and the signal-to-quantisation noise ratio

1. Then the mean squared error due to uniform quantisation is given by $\int_{-\frac{1}{2}}^{\frac{1}{2}} Q^{2} d Q=\frac{1}{12}$.


Figure 3-4: Quantisation noise shown as errors relative to original analogue waveform.
(SQNR). For example, if a sinusoid is sampled with a peak amplitude of $V=2^{n-1}$, then $V^{2}=2^{2 n-2}=2^{2 n} / 8$, giving the SQNR for a full-scale sinusoid as:

$$
\begin{align*}
S Q N R & =10 \log _{10}\left(\frac{2^{2 n}}{\frac{8}{1}}\right)  \tag{3-1}\\
& =10_{12} \log _{10}\left(2^{2 n} \times 1.5\right) \\
& =6.02 n+1.76 \mathrm{~dB} .
\end{align*}
$$

Under these assumptions the maximum SQNR is approximately $6 n \mathrm{~dB}$ : however, two caveats must be considered. First, it is often assumed, incorrectly, that quantisation noise may be treated as random uncorrelated noise. Second, if the analogue signal is not at maximum amplitude, the noise level still remains the same, rendering quantisation noise more apparent.

### 3.1.3 Data-Domain Conversion

Now that the essentially separate processes of sampling and quantisation have been described, it is instructive to examine the complete digital signal processing chain.

### 3.1.3.1 A-D Conversion

An analogue source, $x(t)$, is digitised by the process depicted in the schematic of Figure 3-5. Prior to sampling, $x(t)$ is first band-limited by a low-pass (L-P) anti-aliasing filter, attenuating
components greater than $f_{s} / 2$. A sample-and-hold produces a pulse amplitude modulated (PAM) signal, having continuous amplitude at regular time intervals determined by $f_{s}$. The PAM pulses are then quantised to produce a pulse code modulated (PCM) signal, $x(n)$.


Figure 3-5: Schematic of the processes involved in analogue-to-digital conversion.

### 3.1.3.2 D-A Conversion

$x(n)$ may then be manipulated by DSP algorithms implementing ASP techniques to produce a discrete output signal, $y(n)$. Digital-to-analogue conversion (DAC) reverses the digitisation process, as Figure 3-6. The sample train, $y(n)$, is converted to analogue form, $y(t)$, by first transforming $y(n)$ to stepped voltage levels. Resampling produces another PAM waveform and a low-pass reconstruction filter then returns the time-domain signal $y(t)$.

### 3.1.4 Comparison of Processing Domains

Any DMC must be, firstly, economically viable and, secondly, operationally useful. As DSP is invariably more costly, it is useful to study the advantages presented by the digital approach.

### 3.1.4.1 Disadvantages of Analogue

In an analogue system, a signal can take on any one of a continuous set of values at any point in time. Degradations introduced in each processing section inevitably accumulate, setting an upper limit on the number of sections that can be cascaded. Impairments can be considered in






Figure 3-6: Schematic of the processes involved in digital-to-analogue conversion.
terms of additive noise or distortion, and timing instabilities such as group-delay effects and jitter. These imperfections can never be separated from the original signal.

### 3.1.4.2 Advantages of Digital

A digital signal can have only discrete values at discrete times. Once a signal has been digitised it can, theoretically, be processed indefinitely without further corruption. As DSP only uses multiplies, additions and delays, audio processing can be constructed with mathematical precision. Problems such as component drift can be virtually eliminated.

Time-division multiplexing (TDM) enables many audio signals on different time-slots to share one word-wide data bus as Figure 3-7. Delay elements may be used creatively or for the more subtle restoration of phase coherence in multi-microphone situations. Finally, direct interfacing at each machine boundary preserves signal fidelity until the final D-A conversion.

### 3.2 Digital Audio Processing

A mixing console can be specified by the number of input channels, $n$, and output groups, $m$. Figure 3-8 shows one possible configuration representative of multitrack mix-down.


Figure 3-7: Time Division Multiplexing on a word-wide parallel bus. ${ }^{\text {a }}$
a. LSB represents the least significant bit, and MSB the most significant bit of the sample word.


Figure 3-8: Simplified schematic of a mixing console during mixdown to stereo.

### 3.2.1 Matrix Representation

The group outputs of a DMC can be represented in simplified form (i.e, partially ignoring channel pan) by the following equation. Note, that neither $R$ or $F$ are dependent on $z$ :

$$
\begin{equation*}
\mathbf{Y}(z)=R F H(z) \mathbf{X}(z) \tag{3-2}
\end{equation*}
$$

### 3.2.1.1 Group Output Vector - $\mathbf{Y}(z)$

Equation (3-3) shows the group output vector, $\mathbf{Y}(z)$, where $y_{t}(z)$ is the output from track $t$ of $m$ :

$$
\mathbf{Y}(z)=\left[\begin{array}{c}
y_{1}(z)  \tag{3-3}\\
y_{2}(z) \\
\cdot \\
y_{t}(z) \\
\cdot \\
y_{m}(z)
\end{array}\right]
$$

### 3.2.1.2 Routing Matrix $-R$

$R$, the routing matrix, is shown in Equation (3-4). If $p_{c}$ is the pan coefficient for channel $c$, then $r_{t, c}$ is either $p_{c}, t=1,3, \ldots$ or $\left(1-p_{c}\right), t=2,4, \ldots$ if $c$ is routed to track $t$ :

$$
R=\left[\begin{array}{cccccc}
r_{1,1} & r_{1,2} & \cdot & r_{1, c} & \cdot & r_{1, n}  \tag{3-4}\\
r_{2,1} & r_{2,2} & \cdot & r_{2, c} & \cdot & r_{2, n} \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
r_{t, 1} & r_{t, 2} & \cdot & r_{t, c} & \cdot & r_{t, n} \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
r_{m, 1} & r_{m, 2} & \cdot & r_{m, c} & \cdot & r_{m, n}
\end{array}\right]
$$

### 3.2.1.3 Fader Coefficient Matrix $-F$

The fader coefficient matrix, $F$, of Equation (3-5), denotes the values of the fader coefficients, $f_{C}$ for each channel $c$ :

$$
F=\left[\begin{array}{cccccc}
f_{1} & 0 & . & 0 & . & 0  \tag{3-5}\\
0 & f_{2} & \cdot & 0 & \cdot & 0 \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
0 & 0 & \cdot & f_{c} & \cdot & 0 \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
0 & 0 & . & 0 & \cdot & f_{n}
\end{array}\right]
$$

### 3.2.1.4 Channel Processing Matrix — $H(z)$

Another diagonal matrix, $H(z)$, specifies the processing within each channel:

$$
H(z)=\left[\begin{array}{cccccc}
h_{1}(z) & 0 & \cdot & 0 & \cdot & 0  \tag{3-6}\\
0 & h_{2}(z) & \cdot & 0 & \cdot & 0 \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
0 & 0 & \cdot & h_{c}(z) & \cdot & 0 \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
0 & 0 & \cdot & 0 & \cdot & h_{n}(z)
\end{array}\right]
$$

where $h_{d}(z)$ is the characteristic of $j$ cascaded second-order filter sections in channel $c$ :

$$
\begin{equation*}
h_{c}(z)=\prod_{i=1}^{j} \frac{a_{0_{i, c}}+a_{1_{i, c}} z^{-1}+a_{2_{i, c}} z^{-2}}{1+b_{1_{i, c}} z^{-1}+b_{2_{i, c}} z^{-2}} \tag{3-7}
\end{equation*}
$$

Such bi-quadratic (or bi-quad) sections are usually implemented in the transposed direct form $\Pi$ of Figure 3-9 [McNally, 1981]. Advantages of this second-order topology include less tendency to overflow and minimal limit cycles effects. ${ }^{1}$ Although IIR filters are sensitive to coefficient quantisation, this ordering of DSP primitives is minimally effected.


Figure 3-9: Transposed direct form II, or transposed canonical, bi-quad section.

### 3.2.1.5 Input Vector - $\mathbf{X}(z)$

To complete the matrix representation the input vector, $\mathbf{X}(z)$, contains the input, $x_{0}$ to each console channel $c$ of $n$ :

[^11]\[

\mathbf{X}(z)=\left[$$
\begin{array}{c}
x_{1}(z)  \tag{3-8}\\
x_{2}(z) \\
\cdot \\
x_{c}(z) \\
\cdot \\
\cdot \\
x_{n}(z)
\end{array}
$$\right]
\]

### 3.2.1.6 Summary

If the dimensions of a matrix, $X=\left(x_{i j}\right), 1 \leq i \leq r, 1 \leq j \leq c$, are denoted by $X_{r c}$, then, from matrix algebra, the product $A B=C_{m n}$ is defined only if $A_{m p}$ and $B_{p n} . A_{m p} B_{p n}$ requires $m p n$ multiplications $(\times)$ and $m(1-p) n$ additions $(\Sigma)$, or alternatively $m(1-p) n$ MAC and $m n \times$. If, however, $A$ or $B$ is a diagonal matrix then $m n \times$ alone is sufficient.

From this analysis, implementing $\mathbf{Y}(z)$ in Equation (3-2) requires $n(m+2) \times$ and $m(n-1) \sum$, with an additional $2 n \times$ and $n \sum$ for $R$ and a further $5 j n \times$ and $4 j n \sum$ for $H(z)$. Consequently, a DMC configured with $n=96, j=6, m=24$, and $f_{s}=48 \mathrm{kHz}$ needs a minimum of $2.24 \times 10^{8} \mathrm{MAC} / \mathrm{s}$ and $4.26 \times 10^{7} \times / \mathrm{s}$, equivalent to a nominal 267 MIPS. ${ }^{1}$

### 3.2.2 Processing Redundancy

Recognising that the routing matrix $R$ in Equation (3-4) is normally sparse achieves a major reduction in hardware: every possible input is never routed to every possible output.

### 3.2.2.1 Tracking

When used for tracking, the aim is normally to keep the mix uncommitted by preserving discrete microphone signals on tape. As a result there is a one-to-one correspondence between inputs and outputs, so $n=m$. R now has dimension $R_{m m}$ and the number of DSP operations required to implement this configuration is reduced by a factor of $m / n$ to 67 MIPS.

[^12]\[

R_{tracking}=\left[$$
\begin{array}{cccccc}
r_{1,1} & 0 & \cdot & 0 & \cdot & 0  \tag{3-9}\\
0 & r_{2,2} & \cdot & 0 & \cdot & 0 \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
0 & 0 & \cdot & r_{t, c} & \cdot & 0 \\
\cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\
0 & 0 & \cdot & 0 & \cdot & r_{m, n}
\end{array}
$$\right], n=m
\]

### 3.2.2.2 Mixdown

During mixdown, all channels are routed to two groups as shown in Figure 3-8. ${ }^{1} R$ is sparse with only 2 rows populated and its effective dimension is reduced to $R_{2 n}$. Processing resources required drops to 166 MIPS, some $60 \%$ of nominal. Experience suggests that channel processing is sparsely used, from $0 \%$ on 'classical' to around $30 \%$ for 'pop' work.

$$
R_{\text {mix-down }}=\left[\begin{array}{ccccccc}
r_{1,1} & r_{1,2} & r_{1,3} & \cdot & r_{1, c} & \cdot & r_{1, n}  \tag{3-10}\\
r_{2,1} & r_{2,2} & r_{2,3} & \cdot & r_{2, c} & \cdot & r_{2, n} \\
0 & 0 & 0 & . & 0 & . & 0 \\
. & . & . & . & . & . & . \\
0 & 0 & 0 & . & 0 & . & 0 \\
. & . & . & . & . & . & . \\
0 & 0 & 0 & . & 0 & . & 0
\end{array}\right]
$$

### 3.2.3 Audio Performance

As both consumer media and professional digital production equipment have displayed audible imperfections, it is important to consider the main sources of such problems.

### 3.2.3.1 Sampling Rate \& Frequency Response

The highest fixed sampling rate specified by international standards is 48 kHz . Even at this rate, the construction of low-pass analogue reconstruction filters providing over 100 dB attenuation is extremely difficult. Given that 48 kHz is only 8 kHz higher than a single octave above the extremes of human hearing, the extent of the problem becomes clear.

In oversampling, the DAC input is sampled at 4 or 8 times $f_{s}$, allowing the radical slope to be pulled back into the digital domain. Remaining filter requirements are reduced to

1. The switch matrix for the track monitor and cue mixes are invariably similar to $[S]_{\text {mix }}$ down
that easily delivered by standard analogue design. As the market has grown, quality convertors have become available from Crystal and Motorola among other manufacturers.

### 3.2.3.2 Word-Length \& Dynamic Range

The 96 dB SQNR range provided by true 16 -bit conversion serves well in a single pass situation. Due to noise addition from multiple inputs an absolute minimum of 18 -bits resolution is required for mix functions. In DSP, the major problem is ALU rounding error: results longer than the supported word-length are simply rounded off when stored.

Multiplication of any word-wide coefficient will always yield such a result. Mix data may be multiplied 20 or more times, accumulating rounding degradation with each multiply. There is a general acceptance that the minimum mantissa for professional ASP is 24-bits, giving a theoretical dynamic range of 144 dB for fixed point implementations [Eastty, 1986].

### 3.2.3.3 Real-Time Operation

Opportunities for innovation are irretrievably lost if DMCs merely mimic analogue consoles. It is a simple matter to change ASP coefficients in order to tailor frequency characteristics to orchestral or popular music recording. Consider the channel EQ in Equation (3-7): the response of each bi-quad section is entirely determined by the coefficient sets $a$ and $b$.

Processing delays are frustrating: in professional audio, such latency is completely unacceptable. An ASP throughput lag of only 1 ms can make ordinary studio procedures very difficult [Deutsch, 1982]. If the lag should increase beyond this limit, or vary depending on the type of simultaneous operations, overdubbing and track-bouncing become impossible.

### 3.3 Control and Display Processing

As all housekeeping tasks including mix-down automation are related to the same time reference, normally SMPTE timecode, one central computer system should manage all.

### 3.3.1 Control Rate vs. Audio Rate

A major design choice is how to apportion the decoding of control movements and switch closures between control data and audio sample domains. It is generally held that, for audio
consoles, panel scanning need only be performed at a rate of $100-200 \mathrm{~Hz}$. Consequently, the majority of control data should be processed in the control domain.

This approach leaves only coefficient interpolation for the more costly audio processing environment. The command structure should be able to control up to 128 audio channels, including groups, masters and auxiliaries as found in comprehensive music recording consoles, plus the return of all necessary status and display data.

### 3.3.2 Control \& Display Response

The entire control system must produce apparently instantaneous results: ASP must follow control movement in a natural fashion, with no perceptible lag. Acceptable delay between command initiation and ASP response is in the region of 4 ms [Saunders, 1985]. For mutes and other push-button controls, the impression of immediate response is even more critical.

As the brain superimposes any visual lag, display update must also appear instantaneous [Gulick, 1989]. Initial display response has somewhat more latitude, owing to differences in auditory and visual acuity. Investigations at SSL have determined the maximum acceptable display lag to be in the order of 20 ms [Saunders, 1985].

### 3.3.3 Resolution \& Psychoacoustics

Two kinds of resolution must be considered for a given audio control - static resolution and dynamic resolution. For example, most researchers agree that a listener cannot pinpoint more than 15 discrete stereo positions between hard left and hard right: many put this number at 9 . The higher number safely determines the static resolution requirements of a stereo pan-pot.

If an audio signal is dynamically panned using 15 steps, the jumps between discrete positions are clearly audible. Dynamic resolution requirements of stereo pan are, therefore, much greater than the static requirements. Sufficient dynamic resolution of $2^{6}$ to $2^{8}$ steps must be provided to emulate continuous variability for many functions [Saunders, 1985].

### 3.3.4 Automation

Console automation systems perform many different functions apparently simultaneously. Here, that old chestnut of how much automation is actually useful is reconsidered.

### 3.3.4.1 Reset Automation

It has been normal practice to have a 'lock-out' between studio sessions to ensure that console controls are not inadvertently adjusted. Automation SAVE and RESTORE commands allow sessions to be interlaced with the knowledge that previous settings can be exactly reproduced. To be operationally useful, such commands should take no more than a few seconds. Reset automation thus helps offset production cost by allowing more efficient use of studio time.

### 3.3.4.2 Dynamic Automation

Dynamic automation allows control movements during mix-down to be refined incrementally. Current systems operate at roughly 0.25 to 1.0 SMPTE frame accuracy, storing information for approximately 100 dynamic and 200 momentary events. Total dynamic automation (TDA) extends this approach to embrace all control movements across the console.

An advanced console has 30 variable functions in addition to faders and 70 switch functions in addition to mutes in each channel. TDA achieving control resolution and repeat accuracies similar to 'fader only' dynamic automation will require somewhere in the region of 50 times the processing power and data storage of a standard reset system.

### 3.3.4.3 Control Implications

Beyond the economics of this, the designer must also tackle the control implications. Automation systems have evolved over the years to give the engineer many data editing modes such as RELATIVE, UPDATE, and AUTOTAKEOVER, which depend on dedicated status switches and displays. It is left to the reader to envision a suitably self-explanatory and efficient means of providing such facilities for all control functions on a large DMC.

### 3.4 Surface Ergonomics

Ergonomics, or biotechnology, is the study of the man-machine interface (MMI) especially in terms of physiological, psychological and technological requirements.

### 3.4.1 Overview

As control and display conventions have survived and evolved, analogue console design has achieved a high degree of ergonomic integrity. Two factors are particularly important here.

### 3.4.1.1 Patterns

Pattern recognition allows the operator to gaze across a properly designed control surface and acquire a quick grasp of the general situation. On a standard analogue console, clusters of controls serving the same purpose on different channels are easily identified by their location and physical characteristics [Kvalseth, 1983].

For example, all of the equaliser controls are usually positioned in the same row. The spacing of the controls within the equaliser, the colour-coding of their knobs, and the position and status indication of their switches is consistent. It is these patterns that make it easy to differentiate between an equaliser and a compressor, even on a high density control surface.

### 3.4.1.2 Anomalies

Anomalies are the complement of patterns: the anomaly is that which stands out from the pattern. Anomalies serve as visual hooks, directing the operator's focus to a control that may deserve attention [Pheasant, 1988]. Glancing across the control surface effortlessly detects anomalies such as the presence of an EQ IN lamp.

When the operator focuses on that equaliser, anomalies within its local pattern, such as a mid-range boost control set fully clockwise, instantly convey enough information for the human brain to determine if a closer look is warranted. On closer examination, the scale surrounding the knob provides precise definition of the control's value.

### 3.4.2 Control Layout

If all console controls are within reach, then each is just one action away once relative positions are memorised. Controls that lie beyond reach are quite a different story, however.

### 3.4.2.1 Ideal

For ideal monitoring during mix-down, the engineer will sit centrally between the studio monitors. Controls constantly being adjusted should be placed under his hands and the most important visual indicators straight ahead. Moving out from the centre line are controls requiring occasional adjustment and second priority meters and lights. Everything other than re-adjustment of individual faders can be reached without moving up and down the console.

### 3.4.2.2 In-Line

The vast majority of analogue production consoles follow the 'in-line' philosophy. Perhaps the main drawback of this compromise solution to ergonomic problems has been the exchange of console length for console depth. As the demand for functionality has steadily increased, as many control set sets as can be physically accommodated are provided in the channel strip. In most current layouts, it is possible to reach routing and some EQ controls only by standing up.

### 3.4.3 Display Layout

Display information may be provided around or above the control on some kind of legend, on a VDU screen or both. Every indication must be obvious in meaning and easily interpreted.

### 3.4.3.1 Resolution \& Psychoacoustics

The degree of variability provided in a particular processor must be matched by the display. This is an inherent property of mechanically linked control sets but often overlooked in DMC. If control and audio processing emulate continuous variability and the corresponding display only supports a limited resolution, psychoacoustic phenomena will convince even the most experienced listener that they are hearing 'jumps' where none exist [Pheasant, 1988].

### 3.4.3.2 Definition \& Pattern Recognition

While numeric displays provide excellent discrete definition, pattern recognition is extremely low. Differences between one bar-graph column and another instantly convey more useful information than read-outs of ' 30 ' and ' 80 '. The height of a vertical column meaning quantity is intuitively clear but the best way to convey, say, EQ parameters is not immediately obvious. Except for channel and timecode indicators, the worst option is to display numbers directly.

### 3.4.3.3 VDU Displays

The VDU screen is capable of representing virtually any display and has long been used in fader automation. Two screens will fit within the width of a compact console, Figure 3-10, displaying a huge amount of information - combined or superimposed. For example, the left screen may carry time of day, timecode and machine status, channel levels, and mute flags.

The right screen may show assignable control settings with channel processing shown as a response curve or knob pictorial. During mix-down, all track levels or automated fader
levels could be displayed across both screens together with PPM ballistics. Recent products from SSL Ltd., such as Senaria, and Optimix, use two screens in such a fashion.

### 3.4.4 Long vs. Virtual Surfaces

Ergonomic testing has confirmed that operating a conventional 'long' console requires many eye movements. Very few actions are needed as all controls are effectively dedicated. Assessment of identical tasks on a variety of virtual systems by local expert operators shows that the number of eye movements is surprisingly similar in each case [Stavrou, 1993].

Given sufficient familiarity, the time required for operations is comparable, although observers cannot see the mental gymnastics involved. If the number of controls is reduced the number of actions and keystrokes required is inevitably increased. Ergonomic testing of surface demands confirms that the weak link between operator and DMC is an ergonomic one.

### 3.5 Assignable Mixing

Recognising that the operator can examine and operate only a limited number of controls at one time has brought about the concept of assignability. ${ }^{1}$

### 3.5.1 Overview

The standard way of providing free-grouping in professional consoles is by latching pushbuttons. Besides occupying considerable space, this vast number of switches is a serious cost element. In many designs channel-to-track routing buttons have been centralised into a single assignable panel. Figure 3-10 shows how this concept can be extended across the console.

Processing functions are also centralised: access to a particular channel is made by pressing the 'access' button adjacent to each fader. One set of controls and displays is moved from channel to channel, enabling the engineer to remain in the central listening position. This is clearly better than dodging back and forth to confirm that the stereo image is correct.

[^13]

Figure 3-10: Outline diagram of the assignable console concept.

### 3.5.2 Implementation Issues

A control set may perform the same function for a range of channels or different functions for the same channel. Conventional console design weighs heavily in favour of the later option.

### 3.5.2.1 Per-Channel

Sharing local control sets between different functions provides a means of temporarily 'locking' any desired function. The only requirement is that a sufficient number of parameters be provided to accomplish all necessary control functions. However, this arrangement denies the operator the primary pattern recognition clues normally used to differentiate functions.

### 3.5.2.2 Full-Function Master

An alternative scheme is to provide two or more assignable full-function control sets. Each one has a fixed location and readily identifiable physical characteristics, providing Simultaneous access to one channel's processing. This integration of controls and displays is less physically and mentally taxing than a separated system which splits the operator's focus.

### 3.5.2.3 Control Layout

The proximity of 'per channel' assignable control sets to a channel's 'permanently assigned' controls is a definite plus. On the other hand, the removal of assignable functions from the
channel frees channel real estate for more 'permanently assigned' controls. Control spacing can also be increased to allow the use of displays with greater clarity and definition.

### 3.5.2.4 Comparison

It is not clear which feature is more valuable: the ability to adjust all interacting parameters on a given channel, or certain values of specific functions on different channels simultaneously. One needs to consider how many parameters can be simultaneously controlled and how many 'full-function master' control sets are provided. Substantial differences of opinion exist between mixing engineers with considerable experience and superb credentials [Jones, 1985].

### 3.5.3 Disadvantages of Assignable Faders

The main drawback of assignability is betrayed by its name: access to ASP functions is no longer immediate. This applies not only to control access, but display access as well. While it is not desirable for a DMC control surface to map directly onto a conventional in-line configuration, ergonomic problems in the MMI must still be minimised.

It is generally held that, irrespective of whatever else is assignable, there still has to be a fader dedicated to every input channel. ${ }^{1}$ The more that automated mixdown is used, the less weight this argument holds. Now the channels to be manipulated in one pass of the multi-track can be assigned to a limited number of faders: that number is unlikely to exceed 10 .

### 3.5.4 The Virtual Console

The term 'virtual console' has been loosely applied to any console that separates control and ASP functions. Strictly, the term refers to a DMC architecture that does not, in fact, exist.

### 3.5.4.1 Screen-Based Digital Mixing

One approach to the virtual console is to replace the real control surface by a picture on a screen manipulated by a mouse or similar pointing device. Whilst appearing most ingenious, this approach forces the operator to access controls in a purely serial fashion. Such designs are not usable in real-time and are essentially editing rather than true mixing systems.

[^14]Edit stations have been proposed on which the engineer may prepare control sequences off-line. Unlike video, where still frames have meaning, one can hardly perform audio mixing without auditioning ASP operations in real-time. Clearly, a far more advanced MMI is necessary than that offered by purely screen-based digital console designs.

### 3.5.4.2 A Very Small Console

In the VSC60 virtual surface prototype Stavrou and his colleagues redesigned a tactile control surface from first-principles, discarding many axioms of surface design. To perform the same tasks as conventional consoles needs a radical shift in allocation of human resources. For example, eye movements are reduced to just $0.39 \%$ of operating time [Stavrou, 1993].


Figure 3-11: Outline drawing of the control layout of the VSC60 (courtesy AES Inc.).

At first glance the VSC60 in Figure 3-11 looks like a relatively conventional 10channel desk. Secondary controls such as routing, filters, and dynamics are placed in a selfassigning Ancillary Block. Note the lack of multi-function controls: even from across the control room, traditional visual pattern and anomaly clues are preserved intact.

### 3.6 Conclusions

Recent trends suggest the real possibility of an integrated digital sound production chain between studio microphones and PWM-powered loudspeakers. AES/EBU Serial Transmission Formats (ANSI S4.40-1985), co-axial and fibre-optic consumer standards, and MADI are major steps towards this goal. What is still required is an all-digital studio console.

A vital stage in DMC implementation is the ASP sub-system itself. In this chapter, the author investigates this part from a novel perspective. Not only does this approach determine the level of real-time DSP necessary for a 96:24 DMC (267 MIPS), but it also reveals how to exploit inherent redundancy during tracking ( 67 MIPS) and mix-down ( 166 MIPS).

Much effort has been invested in advancing digital ASP techniques. In contrast, the layout of the control surface, or man-machine interface, is not generally regarded as greatly affecting usability. Freedom from the constraints imposed by mechanical linkage, however, creates a number of potential opportunities to improve the man-machine interface (MMI).

Assignable mixing reduces the graphic density of the control surface and the total number of control sets required for a given configuration. Digital implementation makes TDA a technical reality although at significant cost uplift. In a mature digital architecture, the only restrictions should be the amount of processing resource and the imagination of the engineer.

On a virtual, or fully-assignable, console out-of-reach controls are physically removed: only the correct sequence of keystrokes brings them within 'reach'. Each virtual console uses different sequences of keystrokes to access out-of-reach control sets. It would appear that the MMI is becoming more maladroit with each advance in surface technology.

In Stavrou's VSC60, a selector strip adds to the engineer's orientation, altering this 10 -channel desk ( 220 knobs ) as though the engineer moved to that position. Some multifunction designs require 400 keystrokes to bring all these parameters within reach. Notably, such improvements are solely as a result of surface design and not advanced technology.

At the start of this chapter, Figure 3-1 showed the DMC as consisting of three major sub-systems. The next part of this thesis on digital mixing consoles will concentrate on the third of these. As the logical starting point for any M-P design is the choice of processing element, the next chapter focuses on this specific aspect of the ASP computation engine.

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## Chapter 4

## Programmable DSP Devices

Many semiconductor manufacturers have developed DSP VLSI suitable for digital ASP. Here, the principle features of available processing technologies are analysed and compared.

### 4.1 Introduction

Differences between reduced-instruction-set computers (RISCs), superscalar ${ }^{1} \mu \mathrm{Ps}$, and programmable DSPs result mainly as a consequence of real-time design decisions.

### 4.1.1 History of DSP Devices

True programmable DSPs began to appear 25 years ago with the Bell Labs DSP1. In 1984, Texas Instruments (TI) introduced the popular TMS32010 with a full suite of development software. Motorola, a relative newcomer, shipped its first DSP in 1987. As shown in Figure 4-1, peak performance has doubled approximately every 3 years. Recent developments have included several devices directly supporting multi-DSP topologies (see Appendix A).

### 4.1.2 General-Purpose vs. DSP Processors

DSPs are, in effect, RISCs optimised for the execution of DSP instructions. A relatively large area of silicon is devoted to an array multiplier and a barrel shifter [Nishitani, 1986]. In contrast, these operations are statistically unimportant for G-P programs. Most G-P processors have a limited number of internal registers and lack a fast multiply instruction.

Such restrictions led to the development, firstly, of specialised real-time DSP subsystems, such as the BBC's bit-slice COPAS- ${ }^{2}$ [McNally, 1982], and then programmable VLSI DSPs. Differences to G-P $\mu$ Ps include: dual-bus architectures, DSP-related addressing modes, and serial interfaces to serve AES/EBU and ADC/DAC peripherals.

[^15]

Figure 4-1: Peak performance of commercially available DSP devices.

### 4.1.3 Benchmarks

MIPS comparisons of DSPs have two disadvantages. Firstly, MIPS is dependent on the actual instruction set. Secondly, MIPS can vary widely between different code on the same device. One of the most basic metrics is the multiply and accumulate (MAC) operation, listed in Table 4-1. While small ASP routines of non-trivial program complexity, such as the biquad filter, can be more representative, the MAC determines the maximum rate of DSP algorithms.

### 4.2 Arithmetic

When evaluating programmable DSPs, one of the first features to consider is numeric format. In this section, the characteristics of fixed-point and floating-point arithmetic are compared.

### 4.2.1 Fixed-Point

Considerable attention must be given to the limitations of fixed-point arithmetic. Precision is lost through quantisation arising from both data-domain conversion and multiplication.

### 4.2.1.1 Multiplication and Quantisation

Quantisation errors occur during multiplication since the number of bits required to specify the product is equal to the sum of the number of bits in each operand. If operands have $n$ bits,

| Company | Part No. | Release Date | MAC (ns) | Precision ${ }^{\text {a }}$ |  |
| :---: | :---: | :---: | :---: | :---: | :---: |
|  |  |  |  | Fixed-Point | Floating-Point |
| Analog Devices | ADSP21020 | 1991 | 80 | 32/40 | 32 or 40/40 |
| AT\&T | DSP32C | 1988 | 80 | 16 or 24/40 | 32/40 |
|  | DSP16A | 1988 | 33 | 16/36 | - |
|  | DSP3210 | 1992 | 30 | 16 or 24/40 | 32/40 |
| Motorola | DSP56001 | 1987 | 60 | 24/56 | - |
|  | DSP96002 | 1990 | 60 | 32/64 | 44/96 |
|  | DSP56116 | 1991 | 50 | 16/40 | - |
| TI | TMS320C30 | 1988 | 50 | 16/32 | 32/40 |
|  | TMS320C50 | 1990 | 35 | 16/32 | - |
|  | TMS320C40 | 1991 | 40 | 24/32 | 32/40 |

Table 4-1: DSPs representative of the most important architectural techniques.
a. $x / y$ : where $x$ is the number of bits in the multiplier operands, and $y$ is the number of bits in ALU operands.
both ALU and accumulators must be $2 n$ bits wide to preserve the full product. As only $n$ bits are ultimately stored, the programmer must determine which bits to keep. Storing the lowest order $n$ bits possible without causing overflow maintains maximum operand precision.

### 4.2.1.2 Overflow

Overflow can occur in two ways in a fixed-point DSP. Either an accumulator overflows, producing large magnitude/sign errors, or high-order bits are discarded when it is stored. These effects can be ameliorated by greater precision for operands, additional accumulator headroom, and saturation limiters. There is a direct trade-off between probability of overflow and number precision as variables must be scaled so that overflow is sufficiently unlikely.

### 4.2.2 Floating-Point

Compared with fixed-point, the role of floating-point arithmetic is to identify the best bits in the result to store and at the same time keep track of scaling changes.

### 4.2.2.1 Overview

A floating-point number, $x$, is made up of a mantissa, $M(x)$, and an exponent, $E(x)$, such that:

$$
\begin{equation*}
x=M(x) \times 2^{E(x)} \tag{4-1}
\end{equation*}
$$

Multiplying two floating-point numbers, $x$ and $y$, gives the product, $z$, as:

$$
\begin{equation*}
z=M(x) \times M(y) \times 2^{E(x)+E(y)} \tag{4-2}
\end{equation*}
$$

A floating-point multiplier must contain both a multiplier and an adder for exponents. In the Motorola DSP96002, for example, the floating-point multiplier takes operands with up to 44 bits ( $32 / 11$ ), with the full precision result (64/11) stored in 96-bit registers.

### 4.2.2.2 Renormalisation

When two mantissas are multiplied together, several of the most significant bits in the result may be identical and, hence, redundant. A shift can dispose of these extra bits and the exponent adjusted to track scaling. In virtually all floating-point DSPs, such 'renormalisation' is done automatically in hardware after each arithmetic operation [Motorola, 1989b].

### 4.2.2.3 Dynamic Range

Figure 4-2 shows that the dynamic range of floating-point arithmetic is much greater than that of fixed-point. 24-bit fixed-point approaches an upper bound SQNR of +144 dB whereas floating-point with a 24-bit mantissa achieves the best performance possible with 24-bits. An 8-bit exponent is sufficient to maintain this SQNR over a dynamic range of $\approx 1500 \mathrm{~dB}$.


Figure 4-2: SQNR and dynamic range, shown as a function of signal level.

### 4.3 Memory

Simultaneous memory accesses are also crucial to the performance of DSPs. This requirement is accomplished through a combination of parallel memory banks and rich addressing modes.

### 4.3.1 Parallel Memories

In conventional von Neumann architectures, an instruction and its operands have to be fetched sequentially. Consequently, many DSPs have employed the Harvard architecture shown in Figure 4-3. Two memories and two busses permit instruction and operand fetch to coincide.


Figure 4-3: Basic Harvard architecture, as used in many early DSPs.

### 4.3.1.1 First Modification - Data Stored in Program Memory

The first modification to the basic Harvard architecture (see Figure 4-4) permits data stored in program memory to be used in arithmetic instructions (e.g. AT\&T DSP32C [Fuccio, 1988]). Instructions with two or more operands (i.e. MAC) still require two memory cycles.


Figure 4-4: First modification - data stored in program memory.

### 4.3.1.2 Second Modification - Multi-Ported Data Memory

The second variation, illustrated in Figure 4-5, is to use multi-ported data memory as utilised in the Fujitsu MB86232. Although often perceived as a costly solution, multi-operand instructions do not require operands to be separated into multiple memory banks.


Figure 4-5: Second modification - multi-ported data memory.

### 4.3.1.3 Third Modification - Instruction Cache

In Figure 4-6, an instruction cache supplements program/data memory. When cached instructions are executed, no instruction-fetch is required and so a memory access is freed for a data fetch (e.g. Hitachi 61810). Such low-overhead looping is a common feature today.


Figure 4-6: Third modification - instruction cache.

### 4.3.1.4 Fourth Modification - Third Memory Bank

An obvious next step is to expand the instruction cache until it becomes a third memory bank, as Figure 4-7. This approach is used in the Motorola DSP56001 and DSP96002 to provide simultaneous on-chip fetches of an instruction and two operands. ${ }^{1}$ In contrast, the TI TMS320C30/C40 use an instruction cache to conserve the external memory cycle [TI, 1987]. ${ }^{2}$


Figure 4-7: Fourth modification - third memory bank.

[^16]
### 4.3.1.5 Fifth Modification - Several Memory Banks

Several DSPs achieve high memory bandwidth by accessing parallel memories twice in each instruction cycle. The Hitachi DSPi, on the other hand, uses six memory banks as Figure 4-8. Instructions with three memory operands (e.g. 2 reads, 1 write) use three of the four data memories and program memory. Simultaneous input/output (I/O) can access the fourth bank.


Figure 4-8: Fifth modification - several memory banks.

### 4.3.2 Demand Ratio

Demand ratio is defined as the total number of memory cycles per instruction cycle, summarised in Table 4-2. Devices with the smallest ratio of two require instruction caches in order to execute instructions with two operands in a single cycle. Circular buffer addressing modes are required in order to implement FIR filters in one instruction cycle per tap.

DSPs with a demand ratio of three do not require an instruction cache, but do require circular buffer addressing for delay lines. When the value is four or higher neither circular buffer addressing nor instruction caches are needed. Note, however, that the demand ratios given here are for internal memory: values are usually much lower when memory is off-chip.

### 4.3.3 Internal and External Memory

As DSPs are intended for stand-alone use, most have memory on-chip: there are more subtle reasons, however. Multiple independent memory banks and multi-ported memories are difficult to implement off-chip as the pin count is excessive. Furthermore, the speed penalty of going off-chip increases the system cost by requiring faster memories.

Motorola's DSP96001 has a 32-bit word-length and three memory banks, each with a 32-bit ${ }^{1}$ address, requiring 192 pins for bussing alone if implemented externally. Multiplexing

[^17]| Part No. | No. of <br> Banks | Demand <br> Ratio | Cache | Internal Memory ${ }^{\text {a }}$ | External <br> Memory |
| :--- | :---: | :---: | :---: | ---: | ---: |
| ADSP-21020 | 0 | $(2)$ | 32 W | - | 4 GW P <br> 16 MW P/D |
| DSP32 | 2 | 2 | no | $2 \times 512 \mathrm{~W}$ D RAM <br> 512 W P ROM | 12 KW |
| DSP32C/3210 | 3 | 4 | no | $2 \times 512 \mathrm{~W}$ D RAM <br> 512 W P RAM or <br> 1 KW P ROM | 4 MW |
| DSP56001 | 3 | 3 | no | $2 \times 256 \mathrm{~W}$ D RAM <br> $2 \times 256 \mathrm{~W}$ D ROM <br> 512 W P RAM | $3 \times 64 \mathrm{KW}$ |
| DSP96002 | 3 | 4 | no | $2 \times 512 \mathrm{~W}$ D RAM <br> $2 \times 512 \mathrm{~W}$ <br> 512 W P RAM | $4 \times 4 \mathrm{GW}$ P D |
| TMS320C30/C40 | 3 | 4 | 64 W | $2 \times 1 \mathrm{KW}$ P/D RAM <br> 4 KW P/D ROM | 16 MW |

Table 4-2: Summary of memory systems for preferred DSPs. ${ }^{\text {b }}$
a. P and $D$ represent program and data memory, respectively.
b. W represents 'words', where word-length is dependent on individual DSP numeric formats.
is not an option due to fast cycle times and costly external demultiplexing. Only the more expensive 200-pin version, the DSP96002, carries two physical busses off-chip.

### 4.3.4 Addressing Modes

While parallel memory banks can increase memory bandwidth, one fundamental problem remains. Instructions must specify as many as three simultaneous memory accesses.

### 4.3.4.1 Register-Indirect Addressing

In register-indirect addressing, a set of address registers permits the specification of several addresses in a one-word instruction. Loaded with the address of one word in a data structure (e.g. the first sample in a delay line), all instructions accessing the delay line then reference this register. Since the register bank is small, few bits are required to specify each register.

### 4.3.4.2 Modulo-Mode Addressing

An efficient alternative to shifting data in memory is the circular buffer (Figure 4-9): $m+1$ contiguous memory locations are allocated beginning at $l$. Using modulo-mode addressing, a register reverts to $m$ when it is incremented beyond $l+m$. Motorola DSPs have 24 registers arranged in triplets: the address is held in $r x$, the increment in $n x$, and the buffer length in $m x$.

Two address ALUs perform post-auto-increment and modulo arithmetic, supporting simultaneous modulo addressing for data and coefficients. In order to simplify address ALUs, circular buffers must begin on power-of-two boundaries. If $(m+1) \leq 2^{k}$ but $(m+1)>2^{k-1}$, then the low order $k$-bits of $l$ must be zero. Consequently, $l$ must fall on a multiple of $2^{k}$.


Figure 4-9: Visualisation of a circular buffer of length $m$.

### 4.3.4.3 Other Addressing Modes

Register-direct and modulo addressing modes cannot be used exclusively in ASP. Most DSPs have some form of immediate addressing (operand is part of instruction) and direct addressing (address is part of instruction) as shown in Table 4-3. Other modes include indexed addressing for non-sequential array access, and bit-reversed for re-ordering FFT output vectors.

| Part No. | Immediate | Direct | Indirect |  |  | Bit <br> Reverse |
| :--- | :---: | :---: | :---: | :---: | :---: | :---: |
|  |  |  | No. of <br> registers | Modulo | Index |  |
| ADSP-21020 | yes | yes | 16 | yes | yes | yes |
| DSP32/32C/3210 | int. only | int. only | 15 | no | no | no/yes/yes |
| DSP56001 | yes | yes | 8 | yes | yes | yes |
| DSP96002 | yes | yes | 8 | yes | yes | yes |
| TMS320C30/C40 | yes | 64 KW | 8 | yes | yes | yes |

Table 4-3: Summary of addressing modes for preferred DSPs.

### 4.4 Pipelining

To keep the instruction cycle times of programmable DSPs short, operations are pipelining. DSP internal pipelines and the different programming styles required are examined below.

### 4.4.1 Interlocking

As shown in Figure 4-10, one model of DSP pipelining distinguishes between instruction fetch, decode, operand fetch, and execution stages. ${ }^{1}$ Instruction fetch occurs at the same time as the previous instruction decode and the operand fetch for the instruction before. Overlapping instructions gives the impression that each one finishes before the next begins.


Figure 4-10: Model of the pipelining within a programmable DSP. ${ }^{\text {a }}$
a. Cross-hatched boxes indicate latches, which latch signals once per instruction cycle. PC represents the program counter, controlling which address is fetched from program memory.

With this scheme, contention can occur for internal resources: e.g. fetching two operands from the same memory bank. In interlocking control logic delays ALU execution to resolve the contention. Of the DSPs discussed here, TI processors make the most use of interlocking [TI, 1987]. Accordingly, the internal timing of these devices is rather elaborate.

[^18]
### 4.4.2 Time-Stationary Coding

In time-stationary coding, an instruction specifies the operations that occur simultaneously. Several DSPs, including the Motorola DSP56001, are built around the outline reservation table of Table 4-4 [Lee, 1989]. Each instruction specifies simultaneous operand fetch and execute operations. In essence, the program model is one of parallelism rather than pipelining.

| Hardware Resource | Instruction Cycle |  |  |  |
| :---: | :---: | :---: | :---: | :---: |
|  | $n$ | $n+1$ | $n+2$ | $n+3$ |
| Program memory |  |  |  |  |
| X-data memory |  |  |  |  |
| Y-data memory |  |  |  |  |
| ALU |  |  |  |  |

Table 4-4: Outline of reservation table for DSPs that use time-stationary coding.

Consider the DSP56001 MAC instruction of Figure 4-11: registers $x 0$ and $y 0$ were loaded from memory in a previous instruction; the product of $x 0$ and $y 0$ is added to the $a$ accumulator in the same cycle. Unlike any other device, the DSP56001 has an adder integrated into the multiplier, so that the MAC actually occurs together in the same hardware.

$$
\operatorname{mac} x 0, y 0, a \quad x:(r 0)+, x 0 \quad y:(r 4)-, y 0
$$

Figure 4-11: Example MAC instruction, in Motorola DSP56001 assembly code.

### 4.4.3 Data-Stationary Coding

In data-stationary coding, a single instruction specifies all operations performed on a set of operands. This approach is more natural since the instruction specifies what happens to given data. The main upshot is that the results of an instruction may not be immediately available. AT\&T floating-point DSPs are the most dramatic examples of data-stationary coding.

Fast interrupts are more difficult with data-stationary than with time-stationary coding. For example, the AT\&T floating-point DSPs have a three-cycle quick interrupt. To accomplish this, a second set of pipeline registers 'shadow' the main set, storing the processor state when an interrupt occurs. ${ }^{1}$ Nested interrupts are, of course, not possible [AT\&T, 1991].

### 4.4.4 Comparison of Coding Methods

Time-stationary coding has a number of advantages. Firstly, the timing of a program is clearer. Secondly, interrupts can be much more efficient, since the programmer has explicit control over the pipeline. The Motorola parts have a very fast interrupt, taking exactly two instruction cycles to pull audio samples from an ADC or AES/EBU peripheral into a circular buffer.

Data-stationary is no less efficient than time-stationary coding. However, to specify a multiply, accumulate and store with time-stationary coding requires several instructions. Other operations proceed in parallel specified by the unused operand fields. In data-stationary coding other operations proceed in parallel due to neighbouring instructions.

### 4.4.5 Branching

One difficulty with pipelining is branching: several problems conspire to make it difficult to achieve efficient branching. First, there may not be sufficient time between instruction fetches to decode a branch instruction before the next fetch. Secondly, in conditional branching, the next instruction fetch cannot occur before the ALU condition codes can be tested.

For reasons of efficiency in ASP, it is necessary to use low-overhead looping rather than branch instructions. In Motorola DSPs, any number of instructions may be included inside a loop and loops are interruptible. Another technique used to avoid the inefficiencies associated with conditional branches are conditional instructions [Motorola, 1989a].

### 4.5 Software Development Tools

DSP software development tools have evolved from simple assemblers to the integrated software tools necessary to cope with ever-increasing DSP complexity.

### 4.5.1 Optimising C Compilers

One of the main problems with programmable DSPs still persists: code must be tuned by hand to meet hard real-time constraints. Optimising parallel C compilers exist for the latest generation of floating-point DSPs. Floating-point arithmetic in conjunction with this compiler

[^19]technology removes the need for much programmer-intensive scaling. Regardless of efficiency, these compilers will inevitably be used for large-scale digital ASP applications.

### 4.5.2 Block-Diagram Environments

It is doubtful that good $C$ compilers alone are a complete solution. While not entirely new in the field of DSP, various very high-level 'iconic' development environments have emerged.

### 4.5.2.1 Gabriel

At the University of Berkeley, Lee et al. have developed a block-diagram environment called Gabriel. Real-time constraints, sample rates, recurrences, conditionals, and iteration are managed systematically. Also capable of generating code for multiple processors, Gabriel's basic facilities have been demonstrated for both the DSP56001 and DSP32 [Lee, 1987a].

### 4.5.2 2 Comdisco Signal Processing Worksystem

Comdisco's Signal Processing Worksystem (SPW) allows ASP algorithms to be quickly implemented with minimal software coding. The designer 'wires-up' DSP algorithms chosen from a library of processing blocks, and then compiles and downloads the resultant assembly code. Processing blocks can be manually allocated for multi-DSP platforms.

### 4.5.2.3 IRCAM Signal Processing Workstation

In addition to developing the dual-i860 ISPW, the IRCAM research organisation in Paris has also developed custom applications for this 93.5 MIPS NeXTcube add-in. Animal provides a real-time object-oriented programming environment, while the graphical programming language $M A X$ is a suited to distributed real-time ASP applications such as musical synthesis.

### 4.5.3 Software Simulators

An essential capability almost totally lacking in software simulators is the simulation and debugging of multiple-DSP systems. Motorola is a notable exception: simulator sub-routines can be called from user-written code. Each call emulates the state change of a processor in one clock cycle. A DMC designer planning to use several DSPx600x can write $C$ code that emulates the interconnection of DSPs, shared memory, and other peripheral hardware.

### 4.5.4 Real-Time Operating Systems

SPOX-OS is a real-time C-based operating system for programmable DSPs providing multitasking, memory-management and device-independent I/O. SPOX-MATH, a library of optimised assembly-language routines, minimises the effort required to code a wide-range of DSP applications. While the inevitable consequence of using SPOX is increased overhead, a designer can develop an entire application in C and then hand-optimise time-critical sections.

### 4.5.5 West vs. East

Japanese manufacturers specifically target their DSP devices at a few large manufacturers with large-volume sales. While their designs provide good support for application-specific routines, development tools suffer as a result. In contrast, North American manufacturers target their products at the signal processing community in general. Consequently, these manufacturers must provide high-level language compilers and integrated development tools.

### 4.6 Processing Technology and DMCs

DMCs must be implemented using the technology of the day. In the 1970s, only discrete logic or bit-slice technology could be tailored to the processing demands of professional ASP.

### 4.6.1 Parallel-Bus DSPs

By the late 1970s, many manufacturers had produced 16-bit fixed-point DSPs. Although used for ASP, audio performance was simply not adequate for professional DMCs [Eastty, 1986].

### 4.6.1.1 Fixed-Point

In 1987, Motorola introduced the first 24-bit fixed-point DSP shown in Figure 5-11. The DSP56001 offers a $24 \times 24$-bit multiplier/adder, $2 \times 56$-bit accumulators and a dual Harvard architecture internally. Problems with operand scaling and limit-cycles were greatly reduced. Some products based on bit-slice parts, such as the SSL 01, were superseded overnight.

The SCI port interfaces to serial protocols such as MIDI, and the SSI to data-domain and digital audio protocol converters. An 8-bit wide HOST port is also provided. Time-


Figure 4-12: Block diagram of the Motorola DSP56001 (courtesy [Motorola, 1989a]).
stationary coding coupled with operational concurrency results in efficient coding of stock ASP algorithms. This DSP has proved popular with professional audio manufacturers. ${ }^{1}$

### 4.6.1.2 Floating-Point

Some console designers have opted for IEEE-format floating-point DSPs. From Figure 4-2, 32-bit floating-point arithmetic does not offer increased resolution over 24-bit fixed-point but does remove the need for scaling. Limit cycles are less problematic through the much increased dynamic range of both signal and coefficients alike. However, these advantages are gained at the expense of increased cost and instruction cycle times (see Appendix A).

### 4.6.2 Serial-Bus DSPs

Sony developed the CXD-xxxx serial-bus DSPs for professional audio use. Dedicated devices, such as the CXD-8307 8:1 mix-bus, are complemented by programmable ASP parts.

### 4.6.2.1 Sony CXD-1160

The CXD-1160 has two stereo inputs, one stereo output and contains a multiplier, ALU, onboard control program and serial host port. Processing speed and instruction cycle run at 128-

[^20]times the word clock. This first-generation part can be programmed to perform a wide variety of ASP tasks including: phase inversion, high and low pass filtering, or 3-bands of sweep EQ.

### 4.6.2.2 Sony CXD-2705

The CXD-2705 is a enhanced version of the CXD-1160. A faster processing speed (256-times word-clock), 52-bit internal word-length and two additional stereo outputs support fader level, muting, output routing and stereo $4: 2$ sub-mix functions. Addition hardware is also integrated into the multiplier unit to realise a true MAC instruction [Hingley, 1992].

### 4.6.2.3 Comparison with Parallel-Bus DSPs

Serial-bus devices offer flexible processing and AES/EBU-transparent audio quality at low cost. A DMC signal path can be created by simply cascading devices. ${ }^{1}$ However, chip count rises as a product of the number of inputs, outputs and processing functions. Furthermore, the cumulative pipeline delay inherent in serial-bus DSPs becomes a serious design problem. Conventional parallel-bus processors are more suitable for large-scale DMC implementations.

### 4.6.3 Application-Specific Integrated Circuits

Unlike DSPs, which are fabricated in runs of 100s of thousands at a time, application-specific integrated circuits (ASICs) are generally made in batches of 10-100 thousand.

### 4.6.3.1 Overview

Only the top layers of metal interconnect define circuit function in semi-custom ASICs. In full-custom ASICs, traditionally the exclusive preserve of IC manufacturers, all layers are defined. Silicon compilers accept an ASIC specification as a CAD logic schematic or VHDL description, which is transformed with minimal intervention into a silicon layout. As a result, design engineers with little knowledge of IC manufacture can produce DMC-specific VLSI.

### 4.6.3.2 Design Methodologies

Static-synchronous is the most straightforward, using edge-sensitive, single-phase clocking schemes. Multi-phase static designs use multi-phase level-sensitive clocking, an approach ideally suited to audio ASIC production. In contrast, multi-phase dynamic designs require a

[^21]clock to maintain their state. ASIC vendors rarely have libraries to support this advanced methodology and few, if any, will accept responsibility for such designs [Naish, 1988].

### 4.6.3.3 Advantages

ASICs release DMC designers from the constraints imposed by DSP VLSI manufacturers. Many analogue and digital functions can be implemented on one wafer, with the number format optimised throughout. Testibility features, such as scan paths, can be built-in as an integral part of the design [Trontelj, 1989]. Neve have implemented such a chip-set, using TI libraries, which are now employed in the recently-released Capricorn DMC.

### 4.6.4 Vector Processors

While the Intel i860 was originally developed for graphics processing, its high-speed vector capabilities are being exploited in a wide range of DSP-intensive applications.

### 4.6.4.1 Overview

The Intel i860, Figure 4-13, brings super-computer design concepts to floating-point $\mu$ Ps. Dual-operation instructions use both FPUs increasing the efficiency of IIR filters and FFTs. In the dual-instruction mode, a RISC integer instruction is executed simultaneously. On-chip, the i860 has both instruction and data caches, with a TLB. Direct MIPS comparisons with DSPs, however, can be misleading: other performance factors are just as critical.

### 4.6.4.2 Peak Performance

Intel quotes 100 MFLOPS (peak) for the 50 MHz part: 2-3 times that of most DSP devices [Intel, 1990]. As the i860 relies heavily on pipelining, this rate can only be achieved by fully utilising dual-operation instructions. In non-vector ASP, cache-misses and pipeline-flushes reduce actual performance to a fraction of peak. Most DSPs, by contrast, do not implement pipelining within the ALU, simplifying programming, particularly hand-coding.

### 4.6.4.3 Data Movement and I/O

To keep an ALU continuously supplied with new data, many DSPs support two EMIs with separate address ALUs. In contrast, the i860 provides only one external bus and can only support concurrent accesses when operating out of on-chip cache. Most DSPs provide one or


Figure 4-13: Block diagram of the Intel i860 (courtesy [Intel, 1990b]).
more I/O port, each with a independent DMA controller. The i860 provides no I/O ports and no DMA controller, making it impossible to access and process data concurrently.

### 4.6.4.4 Context-Switching and Determinism

Large register files and multi-stage pipelines limit context-switch response. The i860 requires 81 instruction cycles to save registers before each switch: DSPs can take as few as 18. Many DSPs enable fast interrupts to be serviced in 2 instructions: the i860 requires 24 . Real-time performance of vector processors depends on cache hit-rate. While many DSPs include cache memory, portions can be locked independently to ensure a $100 \%$ cache hit-rate.

### 4.6.4.5 Application to Real-Time ASP

One way to alleviate I/O limitations is to use a DSP front-end processor to handle real-time I/O, and i860s as back-end compute servers. ${ }^{1}$ Vector processors may then be successfully employed in DSP requiring limited real-time I/O - e.g. synthesis. For I/O-intensive

[^22]applications such as digital mixing, however, programmable DSPs offer many benefits. The i860's single EMI and limited I/O also make it cumbersome to use multiple units together.

### 4.6.5 Parallel DSP Devices

Manufacturers continue to increase DSP performance as silicon processing technologies allow. Historically, very little support has been provided for parallel computation.

### 4.6.5.1 Inmos T801 Transputer

Specifically designed for parallelism, the Inmos transputer can perform DSP operations especially when non-time critical. ${ }^{1}$ As shown in Figure $5-12$, a 32-bit time-slicing CPU is integrated with 4 K bytes of RAM and four serial communication links. CPU performance degradation is $\approx 5 \%$ when all four links are saturated. As the T801 has a non-multiplexed EMI, memory access is 1.5 -times that of an equivalent T800 [Inmos, 1989].


Figure 4-14: Block diagram of the Inmos T801 (courtesy [Inmos, 1989]).

[^23]
### 4.6.5.2 Texas Instruments TMS320C40

With conventional DSPs inter-processor communications (IPC) causes device I/O to quickly saturate [Snell, 1989]. DSP manufacturers addressed this problem by providing two EMIs. The DSP96002 and TMS320C30 extend the dual Harvard architecture beyond the silicon. Even with support for dual-channel DMA, IPC still consumes considerable I/O bandwidth.


Figure 4-15: Block diagram of the TI TMS320C40 (courtesy [TI, 1991]).

TI's TMS320C40 (Figure 4-15) goes further: six bidirectional communication ports support glueless IPC. Each port has a dedicated DMA permitting $20 \mathrm{Mbytes} / \mathrm{s}$ of concurrent I/O. Without these ports, the corresponding IPC would have to be squeezed onto the EMIs. An on-chip analysis module provides efficient parallel debugging facilities [TI, 1991].

### 4.6.5.3 Inmos T9000

The first member of a new generation of transputers, the superscalar 50 MHz T 9000 integrates a 200 MIPS 32 -bit CPU with a 25 MFLOPS 64 -bit FPU, 16 Kbytes of cache/conventional memory, and four communications links [Inmos, 1993]. Inmos provides a powerful development toolset, several optimised parallel compiler and real-time kernels.

Four bidirectional serial links and a dedicated virtual channel processor (VCP) support $80 \mathrm{Mbytes} / \mathrm{s}$. The hardware kernel provides an instruction grouper, multiple interrupts and
sub-microsecond context switching. Up to 8 Mbytes of DRAM can be interfaced with zero external logic. T8xx-T9xxx networks can be built using the C100 system protocol convertor.

### 4.6.5.4 Texas Instruments TMS320C80

With a unique MIMD micro-architecture supporting 200 MIPS, the TI TMS320C80, or multimedia video processor (MVP), brings an unprecedented level of performance to programmable DSP. Four 64-bit DSPs are integrated on a single IC together with a 32-bit RISC CPU, a DMA controller, and 50 Kbytes of static RAM [Andrews, 1994].

With $400 \mathrm{Mbytes} / \mathrm{s}$ IPC bandwidth, the MVP can support real-time applications which currently require large M-P configurations. Although targeted at video processing, the MVP is ideally suited to other high-end applications including large-scale digital audio mixing. TI expects full production to begin by mid 1995 , at a unit cost of $\$ 250$ in 10,000 -unit quantities.

### 4.6.5.5 Philips Semiconductor TriMedia

With patents including the CD , Philips has an impressive pedigree in audio innovation. Now this company has released the TriMedia, a 100 MHz DSP targeted at multimedia [Hars, 1995]. This device employs a VLIW architecture to execute five operations in a single clock cycle. Each operation can contain 11 RISC instructions, making for some impressive benchmarks.

Unlike RISC where the CPU reveals parallelism, VLIW relies heavily on compiler technology. By incorporating serial audio DMA, Philips envisage that the TriMedia will find use in applications requiring multiple DSPs, i.e. audio mixing. With sampling beginning in early 1996, full production of the TriMedia is scheduled for the middle of the year.

### 4.7 Conclusions

Programmable DSPs come in two basic types, fixed- and floating-point. Floating-point DSPs are more expensive and generally at least $50 \%$ slower than fixed-point devices of comparable technology. The main advantage of floating-point devices is that they free the DMC designer from concerns about scaling: the chief disadvantages are reduced speed and increased cost.

The dominant trend in DSPs is towards complexity: each new device includes $\mu \mathrm{P}$ facilities lacking in the previous product. Conversely, a market still exists for simple and fast

DSPs. High-speed. low-power fixed-point DSPs, such as the AT\&T DSP16A and TI TMS320C50, are targeted specifically at mobile communications applications.

An interesting development is the emergence of DSP cores: programmable DSPs surrounded by customer-specified circuitry. For example, Motorola has released the DSP56116, a 16-bit core version of the original 24-bit DSP56001. Devices such as these, and the recent TASP family from Texas Instruments, provide strong competition for ASICs.

A fourth approach to pipelining, not yet implemented in any commercial DSP, is pipeline interleaving. Instead of a single in-line instruction stream, instructions from several independent streams are interleaved. Pipeline interleaving transforms a single DSP with datastationary code into multiple processor slices sharing the same hardware.

With the newest versions of G-P $\mu$ Ps already incorporating DSP-like dual-bus architectures, the obvious next step is the integration of an array multiplier and a barrel shifter. New VLSI technologies such as gallium arsenide (GaAs) suggest that another order-ofmagnitude speed improvement is possible through processing technology alone.

Concepts borrowed from parallel computation have moved across into the DSP arena. Since VLSI devices are subject to stringent packaging restrictions, byte-wide communication links providing highly efficient parallel processing capabilities. Devices, such as the TI TMS320C40, integrate multiple IPC ports directly on-chip with high-speed DSP cores.

While recent advances support M-Ps, few devices have specific architectural features for DMC systems on-chip. For example, the time-domain effects of Section 2.1.3 require dedicated delay processing resources. Although many DSPs include sine-wave tables, none support the additional modulation waveforms necessary for chorus and pitch-shift algorithms.

In order to implement the mix function described in Section 3.2.1, many instruction cycles are consumed by interpolating real-time control parameters in order to remove zipper noise. Incidentally, built-in data-domain converters are not desirable as manufacturers prefer to choose parts from a variety of sources or implement proprietry solutions.

As device technology advances in to the multimedia era, the distinctions between DSP devices and G-P processors are being swiftly eroded. Novel devices like the TMS320C80 are
emerging with multiple DSP cores integrated on a single die. Although currently expensive, their cost compares favourably with the original pricing structure of the Motorola DSP56001.

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## Chapter 5

## Architectural Issues

Due to fundamental physical limits on processor technology, any DMC design demands multi-processor (M-P) implementation. Some form of inter-processor communications (IPC) network is necessary to exchange audio samples and mix-function control parameters.

### 5.1 Terminology

In this section the terminology of parallel processing is developed in the context of supporting computationally intensive real-time ASP on multi-DSP architectures.

### 5.1.1 Classification of Parallel Architectures

Four general classes of parallel-machine architectures are distinguished in the most widely known classification, given by Flynn [Flynn, 1966]. His taxonomy categorises computer systems according to the relationship between instruction and data streams.

### 5.1.1.1 Single Instruction, Single Data Stream (SISD)

In SISD machines there is a single instruction stream and, hence, a single processing unit. Each instruction results in one arithmetic operation on one set of data at a time. This category includes conventional sequential uni-processors such as the Motorola MC68040 $\mu \mathrm{P}$.

### 5.1.1.2 Single Instruction, Multiple Data Stream (SIMD)

SIMD machines have a single master unit, which makes decisions on the flow of control, and a large number of slave processors. As they are designed to exploit data parallelism, SIMD M-Ps retain the single instruction stream but support instructions which execute on many data streams. Process parallelism cannot be utilised here as it demands multiple control threads. ${ }^{1}$

[^24]As a result, SIMD machines perform poorly in applications which contain substantial proportions of sequential code. The most significant machines in this class are array processors such as Active Memory Technology's DAP, and the CM-1 and CM-2 from Thinking Machines which connect 64 K 1-bit PEs in pseudo-hypercube topologies.

### 5.1.1.3 Multiple Instruction, Single Data Stream (MISD)

Though Flynn's taxonomy reveals that MISD architectures might exist, there are currently no examples of machines which apply multiple instructions at once to a single data stream.

### 5.1.1.4 Multiple Instruction, Multiple Data Stream (MIMD)

The multiple instruction streams of MIMD machines lead to a number of devices each with one or more data streams. Processors are either synchronous, performing successive instructions in lock-step, or achieve synchronisation via messages-passing. Note that the interaction of instruction streams can lead to dead-lock and other such software ailments.

Two broad sub-categories of MIMD exist: homogeneous machines constructed from identical PEs, and heterogeneous designs. A large number of commercial M-Ps adhere to the MIMD model, including the Ametek 2010 2-D mesh and Intel iPSC hypercubes. MIMDs using programmable DSP devices are particularly attractive for DMC implementations.

### 5.1.2 Processing Budget

Each channel in a SSL G Series console (see Section 2.3.1) has 78 switch and 31 continuously variable functions, or over 10,000 functions on a 96 -channel frame. In an ideal M-P with $n$ processors, MIPS is increased by a factor of $n /(n-1)$ for each additional PE - linear speed-up. If each additional DSP contributes only $x$-times current performance, then:

$$
\begin{align*}
& S(1)=1 \\
& S(n)=\left(\frac{n-1+x}{n-1}\right) S(n-1), \quad 0 \leq x \leq 1 \tag{5-1}
\end{align*}
$$

where $S(n)$ is the speed-up of $n$ processors. Such diminishing returns, shown in Figure 5-1, make successive DSP less and less cost effective. From Section 3.2.1, the core ASP engine of a 96:24 DMC requires 267 MIPS. Allowing 30 MIPS for additional mix circuits and assuming $x=0.75$, Equation (5-1) implies 80 DSPs with a nominal 10 MIPS.


Figure 5-1: Example speed-up curves for M-P DMC implementations.

### 5.1.3 Implementing Parallelism

There are two fundamental ways in which DSPs can be composed to create M-P architectures. Perhaps the simplest way to introduce parallelism is to replicate a component $n$ times, as shown in Figure 5-2. To exploit this form of parallelism, digital audio streams must be separated in space. For this reason this approach is known as spatial or replicated parallelism.


Figure 5-2: Exploiting spatial parallelism.

Alternatively, ASP can be partitioned into a number of steps, as shown in Figure 5-3. Each step is applied sequentially to restore the original ASP algorithm. Accordingly, this form of parallelism is also known as temporal parallelism. Although the distinction between these forms of parallelism is clear, an increasing number of M-Ps employ both techniques.


Figure 5-3: Exploiting temporal parallelism.

### 5.1.4 Processor Pipelining

Pipelining is an implementation of temporal parallelism whereby multiple tasks are overlapped in time. In a processor pipeline the ASP required to complete an application is broken into smaller tasks, which are then connected one to the next to form a pipe. Samples enter at one end, are processed through the pipe, and exit at the other end, as Figure 5-4.

In DMCs response is perceived as immediate if the system operates with an accumulated processing lag of less than 1 ms . To meet the frequency response requirements of Section 3.1.1.3, each DSP must capture input samples every $20.8 \mu \mathrm{~s}$. In a DMC, this corresponds to a maximum processor pipeline of 48 stages between inputs and outputs. ${ }^{1}$


Figure 5-4: Detail of overlapped processing in a $n$-stage linear pipeline.

[^25]
### 5.2 Multi-Processor Topologies

Over the years a plethora of interconnection networks have appeared in the research literature [Bhuyan, 1989]. Here, the differences among the more exotic topologies are investigated.

### 5.2.1 Single-Stage Networks

Single-stage topologies have particular intrinsic value as they present a uniform methodology to compare and contrast possible M-P configurations for a DMC implementation.

### 5.2.1.1 Chordal Ring

The chordal ring, Figure 5-5, introduces cross-links between nodes on opposite sides of a simple ring. These chordal links reduce the maximum number of links that must be traversed to reach a destination node, and increase tolerance to node and link failure. Optimal choice of choral link placement reduces the diameter of an $n$ node ring from $O(n)$ to $O\left(n^{1 / 2}\right)$.


Figure 5-5: Illustration of chordal ring with 6 nodes.

### 5.2.1.2 X-Tree

The x-tree or full-ring tree is an adaptation of the simple $b$-ary tree, where $b$ is the fanout at each level. All nodes at each level are also connected in a ring, as shown in Figure 5-6. Increasing resilience to faults, this approach also reduces the communication bottleneck near the tree root. Topology diameter is also reduced for binary and ternary configurations.


Figure 5-6: Illustration of 2-ary x-tree with 4 levels.

### 5.2.1.3 Nearest-Neighbour Mesh

With $w$ nodes in each of $d$ dimensions the nearest-neighbour mesh, shown in Figure 5-3 adheres to the generalised d-dimensional hypercube topology. Each of the $w^{d}$ nodes, except peripheral nodes, is connected to 2 nodes in each dimension by point-to-point links. This reduces the number of links by $w^{d-1}$ giving the total number of links as $(w-1) d w^{d-1}$.


Figure 5-7: Illustration of 2-D nearest-neighbour mesh of width 4.

### 5.2.1.4 Spanning-Bus Hypercube

The spanning bus hypercube is a $d$-dimensional lattice of width $w$ in each dimension. Each node of Figure 5-7 is connected to $d$ buses, one in each orthogonal dimension: $w$ nodes share each bus. Note that the binary hypercube used in many of the current generation of commercial M-Ps is a degenerate case of this topology, arising when $w=2 .{ }^{1}$

[^26]

Figure 5-8: Illustration of 3-D spanning bus hypercube of width 3.

### 5.2.1.5 Dual-Bus Hypercube

Pruning $d-2$ connections from each node gives the dual bus hypercube of Figure 5-9. All nodes are connected to a ' 0 -th' dimension, with a second connection in one of the $d-1$ orthogonal hyperplanes. The direction of this bus differs from hyperplane to hyperplane, but repeats if the width $w$ of that hypercube dimension exceeds $d-2$.


Figure 5-9: Illustration of 3-D dual bus hypercube of width 3.

### 5.2.1.6 Torus

Identical to the spanning bus hypercube, the torus replaces the bus connecting each group of $w$ nodes with a ring of point-to-point connections, as shown in Figure 5-10. As a special case, the torus includes the closed nearest-neighbour mesh. Lattices, and hence tori, of higher degree can be obtained by further connections to nearest-neighbour nodes.


Figure 5-10: Illustration of 2-D torus of width 4.

### 5.2.1.7 $R$-ary $N$-cube

The $R$-ary $N$-cube is a generalisation of the indirect binary $n$-cube or butterfly network. As there are $N$ levels and $R^{N}$ nodes on each level, the network contains $N R^{N}$ nodes each connected to $2 R$ other nodes and therefore $N R^{N+1}$ links. Unlike Pease's original proposal, the cube is closed: the top and bottom rows shown in Figure 5-11 are, in fact, the same.


Figure 5-11: Illustration of 2-ary 3-cube.

### 5.2.1.8 Binary Hypercube

A binary hypercube of dimension $d$ has $2^{d}$ nodes, each connected to one node in each $d$ orthogonal dimensions: a total of $d 2^{d-1}$ links. Figure 5-12 shows the recursive creation of hypercubes from a single node which is, by definition, a 0-dimensional hypercube. Mapped onto an $n$-dimensional co-ordinate system, the label of a node is then a tuple of $n$ binary digits.

### 5.2.1.9 Cube-Connected Cycles

Also proposed to limit the number of connections, cube-connected cycles replace each $2^{d}$ nodes of a binary hypercube with a ring of $d$ nodes. Each ring of Figure 5-13 connects to one of $d$ links incident on the vertex. A fixed node valency of three gives a total of $3 d 2^{d-1}$ links. Expanding the dimension $d$ requires $(d+2) 2^{d}$ additional nodes and $3(d+2) 2^{d-1}$ links.


Figure 5-12: Recursive creation of binary hypercubes of dimension $d$.


Figure 5-13: Illustration of 3-D cube-connected cycles.

### 5.2.2 Analysing M-P Topologies

In the following sections $x$-tree, mesh, torus and binary hypercubes are used to illustrate the analytic techniques used to construct the Topology Analysis Tool detailed in Appendix C.

### 5.2.2.1 X-Tree

If $b$ is the fanout at each level $l$ in a $b$-ary tree, then the number of nodes is a geometric series:

$$
\begin{equation*}
n=\sum_{x=1}^{l} b^{x-1}=\sum_{x=0}^{l-1} b^{x}=\frac{b^{l}-1}{b-1} \tag{5-2}
\end{equation*}
$$

Each node is connected to the level above, to $b$ nodes in the level below, and all nodes at each level are connected in a ring, as Figure 5-6. With the node valency fixed at $b+3$, the number of connections is $(b+3) n$ less $b^{l}-3$ missing for peripheral nodes, giving $2(n-1)$ links.

Expanding the x-tree topology by incrementing $b$ requires $\left((b+1)^{l}-b^{l}-n\right) / b$ nodes whereas incrementing $l$ requires $b^{l}$. Since the number of links is directly proportional to $n$, it increments by $2 n$ in both cases. Topology diameter is $\max (2(l-2), l-1)$ for $b=2$, $\max (2 l-3, l-1)$ for $b=3$ and then $2(l-1)$ for $b \geq 4$ as in a conventional $b$-ary tree.

### 5.2.2.2 Nearest-Neighbour Mesh

If all $w^{d}$ nodes in a mesh were connected to 2 nodes in each dimension there would be a total of $2 d w^{d}$ connections to $d w^{d}$ links as in the torus below. As noted in Section 5.2.1.3, opening the nearest-neighbour mesh reduces the number of links to $(w-1) d w^{d-1}$. With a uniform node valency of $2 d$, unused connections are then available for peripheral I/O devices.

Incrementing the width of the mesh $w$ increases the number of nodes by $(w+1)^{d}-w^{d}$, whereas incrementing the dimension $d$ increases $n$ by $(w-1) w^{d}$. The corresponding increases in the number of links are $w d(w+1)^{d-1}-(w-1) d w^{d-1}$ and $(w-1) w^{d-1}[w(d+1)-d]$ respectively. The maximum inter-node distance is $(w-1) d$.

### 5.2.2.3 Torus

The $d$-dimensional torus, a $d$-dimensional mesh of width $w$, connects each of $w^{d}$ nodes to a ring of size $w$ in each of $d$ orthogonal dimensions (see Figure 5-10). As each node is connected to $d$ rings, there are a total of $2 d w^{d}$ connections to $d w^{d}$ links. Extension of results for the binary hypercube leads to link and node connectivities of $2 d$ for arbitrary $w$.

Topology diameter is $d\lfloor w / 2\rfloor$, that is $d$ times the maximum inter-node distance in any dimension using that dimension's ring. The torus, with its lattice structure, has node expansion increments of $(w+1)^{d}-w^{d}$ increasing $w$, and $(w-1) w^{d}$ increasing $d$. Since expanding $d$ requires an increase in valency, the later usually requires the M-P to be re-wired.

### 5.2.2.4 Binary Hypercube

The binary hypercube, Figure 5-12, contains $2^{d}$ nodes each connected to a single node in each of $d$ orthogonal dimensions. A special case of the spanning-bus hypercube, this topology is also a degenerate torus, obtained when $w=2$ and the 2 -node ring replaced with a single link. Hence, there are $d 2^{d}$ link connections and $d 2^{d-1}$ communication links.

As adjacent node labels are identical except in one bit position, the topology diameter is thus $d$. As was noted in Section 5.2.1.8, a $d+1$ dimensional hypercube is constructed recursively from interconnecting two identical $d$-dimensional hypercubes. Consequently, the node expansion increment of a $d$-dimensional binary hypercube is simply $2^{d}$.

### 5.3 Topology Metrics

Many configurations have been proposed for M-P architectures. Many topologies have resulted in actual practical M-P designs while the rest serve as theoretical models.

### 5.3.1 Overview

Graphs are probably the most natural model to describe M-P INs. Each graph consists of $n$ nodes and a set of point-to-point links connecting pairs of nodes, as Table 5-1. Physically, each node corresponds to one compute processing element (PE). It is possible to define a set of metrics which provide a framework to systematically compare alternative INs. In the following sections, these metrics are defined and then determined for various M-P topologies.

### 5.3.2 Expansion Metrics

A fundamental consideration is the effect of altering the degree of replication, usually upward as the end-user runs out of compute power. Ideally, a DMC should be incrementally expandible: however, scalability may be restricted by topology-dependent factors. Table B-1 shows that the smallest increments for many M-Ps are dependent on the current configuration.

As M-P topologies are not necessarily incrementally expandible, so the link expansion metrics detailed in Table B-2 are dependent on both $n$ and the specific topology. As the degree of replication is large then a careful analysis must be performed. However, the ability to extend a particular multi-DSP console is clearly only important within a given range. ${ }^{1}$

[^27]| Topology | Number <br> of Nodes | Number of <br> Connections | Number of <br> Links |
| :--- | :---: | :---: | :---: |
| full interconnection | $n$ | $n(n-1)$ | $\frac{n(n-1)}{2}$ |
| shared bus | $n$ | $n$ | 1 |
| simple ring | $n$ | $2 n$ | $n$ |
| chordal ring | $\frac{b^{l}-1}{b-1}$ | $2(n-1)$ | $n-1$ |
| $b$-ary tree | $\frac{b^{l}-1}{2}$ | $4(n-1)$ | $2(n-1)$ |
| $b$-ary x-tree | $n^{l}$ | $\frac{3 n}{2}$ | $2(w-1) d w^{d-1}$ |
| nearest-neighbour mesh | $w^{d}$ | $2(w-1) d w^{d-1}$ |  |
| spanning-bus hypercube | $w^{d}$ | $d w^{d}$ | $d w^{d-1}$ |
| dual-bus hypercube | $w^{d}$ | $2 w^{d}$ | $2 w^{d-1}$ |
| torus | $w^{d}$ | $2 d w^{d}$ | $d w^{d}$ |
| $R$-ary $N$-cube | $N R^{N}$ | $2 N R^{N+1}$ | $N R^{N+1}$ |
| binary hypercube | $2^{d}$ | $d 2^{d}$ | $d 2^{d-1}$ |
| cube-connected cycles | $d 2^{d}$ | $3 d 2^{d}$ | $3 d 2^{d-1}$ |

Table 5-1: M-P topology physical characteristics.

### 5.3.3 Network Connectivity

A DMC should continue to function with reduced capacity when components fail. This property is particularly important for large M-Ps as the probability of flawless operation decreases as $n \rightarrow \infty$, for constant MTBF. The link connectivity $l c(i, j)$ between two nodes $i$. and $j$ is defined as the minimum number of links that must be removed to disconnect $i$ and $j$.

Similarly, node connectivity $n c(i, j)$ is defined as the minimum number of nodes other than $i$ or $j$ that must be removed to disconnect $i$ and $j$. Network connectivity metrics therefore
measure the resiliency of a M-P to faults and subsequent ability to continue operation. As
Table 5-2 illustrates, there is wide variation in the connectivities of M-P topologies.

| Topology | Number of Nodes | Link Connectivity | Node Connectivity |
| :---: | :---: | :---: | :---: |
| full interconnection | $n$ | $n-1$ | $n-1$ |
| shared bus | $n$ | 1 | $n$ |
| simple ring | $n$ | 2 | 2 |
| chordal ring | $n$ | 3 | 3 |
| $b$-ary tree | $\frac{b^{l}-1}{b-1}$ | 1 | 1 |
| $b$-ary x-tree | $\frac{b^{l}-1}{b-1}$ | 3 | 3 |
| nearest-neighbour mesh | $w^{\text {d }}$ | $d$ | $d$ |
| spanning-bus hypercube | $w^{\text {d }}$ | $d$ | $d(w-1)$ |
| dual-bus hypercube | $w^{\text {d }}$ | 2 | $2(w+1)$ |
| torus | $w^{d}$ | $2 d$ | $2 d$ |
| $R$-ary $N$-cube | $N R^{N}$ | $2 R$ | $2 R$ |
| binary hypercube | $2^{d}$ | $d$ | $d$ |
| cube-connected cycles | $d 2^{d}$ | 3 | 3 |

Table 5-2: M-P topology network connectivities.

### 5.3.4 Topology Diameter

The maximum internode distance, or topology diameter $\delta$, places a lower bound on the delay required to propagate data. It is defined as the maximum number of IPC links that must be traversed to pass a message to any node along a shortest path [Fleckenstein, 1992]. Topologies with relatively large $n$ which still maintain small $\delta$ are often referred to as dense networks.

From Table 5-3, the diameter of most M-P topologies is $O\left(\log _{p} n\right)$ or $O\left(n^{1 / p}\right)$ where $p$ is a topology specific parameter. In isolation, the diameter alone is not a good general measure
of a network's communication capacity as it is biased towards topologies constructed using buses: a shared bus has the same maximum internode distance as full interconnection.

| Topology | Number of Nodes | Topology Diameter, $\delta$ | Order of $\delta$ |
| :---: | :---: | :---: | :---: |
| full interconnection | $n$ | 1 | 1 |
| shared bus | $n$ | 1 | 1 |
| simple ring | $n$ | $\left\lfloor\begin{array}{l}n \\ 2\end{array}\right\rfloor$ | $n$ |
| chordal ring | $n$ | $\left\lceil\begin{array}{l}n \\ \frac{4}{4}\end{array}\right.$ | $\sqrt{n}$ |
| $b$-ary tree | $\frac{b^{l}-1}{b-1}$ | $2(l-1)$ | $\log _{b} n$ |
| $b$-ary x-tree | $\frac{b^{l}-1}{b-1}$ | $\max (2(l-\max (3-b / 2,1)), l-1)$ | $\log _{b} n$ |
| nearest-neighbour mesh | $w^{d}$ | $(w-1) d$ | $\log _{w} n$ |
| spanning-bus hypercube | $w^{\text {d }}$ | $d$ | $\log _{w} n$ |
| dual-bus hypercube | $w^{d}$ | $2(d-1)$ | $\log _{w} n$ |
| torus | $w^{\text {d }}$ | $d\left\lfloor\frac{w}{2}\right\rfloor$ | $w \log _{w} n$ |
| $R$-ary $N$-cube | $N R^{N}$ | $N+\left\lfloor\begin{array}{l}N \\ 2\end{array}\right]$ | $\log _{R} n$ |
| binary hypercube | $2^{\text {d }}$ | d | $\log _{2} n$ |
| cube-connected cycles | $d 2^{d}$ | $2 d+\left\lfloor\frac{d}{2}\right\rfloor-1$ | $\log _{2} n$ |

Table 5-3: M-P topology diameters.

### 5.3.5 Node Valency

Table 5-4 details node valencies $v$ for the M-P topologies considered here. Unfortunately, technological constraints such as pin-out typically limit $v \leq 6$. While some topologies scale independent of $v$, many rely on $v$ scaling with a topology-specific parameter. Often VLSIs
support a constant off-chip bandwidth $\beta$ which must support the $v$ IPC ports. From Table 5-1 it is clear that node valency has a direct bearing on the connectivity metrics of M-P topologies.

| Topology | Number <br> of Nodes | Valency, $v$ |
| :--- | :---: | :---: |
| full interconnection | $n$ | $n-1$ |
| shared bus | $n$ | 1 |
| simple ring | $n$ | 2 |
| chordal ring | $\frac{b^{l}-1}{b-1}$ | 3 |
| $b$-ary tree | $b^{l}-1$ |  |
| $b-1$ | $b+1$ |  |
| $b$-ary x-tree | $b+3$ |  |
| nearest-neighbour mesh | $w^{d}$ | $2 d$ |
| spanning-bus hypercube | $w^{d}$ | $d$ |
| dual-bus hypercube | $w^{d}$ | $d$ |
| torus | $w^{d}$ | $2 d$ |
| $R$-ary $N$-cube | $N R^{N}$ | $2 R$ |
| binary hypercube | $2^{d}$ | $d$ |
| cube-connected cycles | $d 2^{d}$ | 3 |

Table 5-4: M-P topology node valencies.

### 5.3.6 Diameter.Valency Product

As can be seen from Table 5-4 and 5-5, topology diameter $\delta$ is mitigated by node valency $v$. In open networks, edge processors require fewer links and dangling links are available for I/O. In homogeneous closed M-Ps, all PEs have equal $v$ and I/O must be accomplished by breaking a link or through the use of supplementary hardware. To quantify this interdependency, $\delta \cdot u$ describes the system costs incurred as a consequence of minimising $\delta$.

| Topology | Number of Nodes | Diameter.Valency Product, $\delta \cdot \mathrm{v}$ |
| :---: | :---: | :---: |
| full interconnection | $n$ | $n-1$ |
| shared bus | $n$ | 1 |
| simple ring | $n$ | $2\left\lfloor\begin{array}{l}n \\ \frac{2}{2}\end{array}\right]$ |
| chordal ring | $n$ | $3\left\lceil\begin{array}{l}n \\ 4\end{array}\right.$ |
| $b$-ary tree | $\frac{b^{l}-1}{b-1}$ | $2(b+1)(l-1)$ |
| $b$-ary x-tree | $\frac{b^{l}-1}{b-1}$ | $(b+3) \max (2(l-\max (3-b / 2,1)), l-1)$ |
| nearest-neighbour mesh | $w^{\text {d }}$ | $2(w-1) d^{2}$ |
| spanning-bus hypercube | $w^{\text {d }}$ | $d^{2}$ |
| dual-bus hypercube | $w^{\text {d }}$ | $2 d(d-1)$ |
| torus | $w^{\text {d }}$ | $2\left\lfloor\frac{w}{2}\right\rfloor d^{2}$ |
| $R$-ary $N$-cube | $N R^{N}$ | $2 R\left(N+\left\lfloor\frac{N}{2}\right\rfloor\right)$ |
| binary hypercube | $2^{d}$ | $d^{2}$ |
| cube-connected cycles | $d 2^{d}$ | $3\left(2 d+\left\lfloor\frac{d}{2}\right\rfloor-1\right)$ |

Table 5-5: M-P topology diameter.valency products.

### 5.4 Comparison of M-P Topologies

Here, the metrics defined in the previous section and the Topology Analysis Tool are applied to various M-P topologies in order to determine the optimum connectivity subset for DMCs.

### 5.4.1 Simple Topologies

In the context of this comparison, 'simple' includes shared-bus and ring topologies. The number of connections and links are linearly dependent on $n$, except that a shared-bus has, by
construction, only one IPC 'link'. All three topologies have a node increment of 1 , though a chordal-ring with odd $n$ is degenerate as one PE cannot connect to a cross-topology link.

One additional link is required in a simple-ring, whereas chordal-rings expand by 3 links for every 2 PEs. Network connectivities equal $v$, except the node connectivity of a shared-bus which must be $n$. Due to cross-topology links, $\delta$ for a chordal-ring is half that of the simple equivalent. Valencies of 1,2 , and 3 give the $\delta \cdot v$ characteristics of Figure 5-14.


Figure 5-14: $\delta \cdot u$ characteristics for simple topologies.

### 5.4.2 Tree Topologies

From Table 5-1, the number of point-to-point links required in a simple $b$-ary tree is $n-1$. Due to the ring at each level, the number required in an x-tree is twice that of the simple tree.

### 5.4.2.1 Network Connectivity

From a practical stand-point, it is poor DMC design to expand a tree-based M-P by incrementing $b$ as the PE valency does not remain fixed. Incrementing $l$, the node increment for expanding any $b$-ary tree is linear with respect to $n$, i.e. $n(b-1)+1$. From Figure 5-15, the same is true for link increments for simple $b$-ary topologies but is 2-times this factor for similar x-trees. As a result, expansion factors for both topologies converge to $b$ as $n \rightarrow \infty$.


Figure 5-15: Link increments for tree topologies.

### 5.4.2.2 Diameter \& Valency

Since the valency of $b$-ary topologies is $b+1$ for simple trees and $b+3$ for x -trees, link and node connectivities are simply 1 and 3, respectively. As noted in Section 5.2.1.2, 2-ary and 3 -ary x-trees have a definite $\delta$ advantage over their simple $b$-ary counter-parts, giving the $\delta \cdot v$ characteristics of Figure 5-16 below. Because of the practical limits on PE valency discussed in Section 5.3.5, the $b+3$-valent 4-ary x-tree is not considered further in this thesis.


Figure 5-16: $\delta \cdot v$ characteristics of tree topologies.

### 5.4.3 Hypercubes

In this section, the hypercube topologies discussed previously are evaluated. 4-D mesh and 4-D torus implementations are excluded from this analysis as they have a node valency of 8 .

### 5.4.3.1 Construction Techniques

While a 2-D mesh requires as few connections as other hypercubes for small $n$, this is not the case for the size of M-P required for a DMC system. Dual-bus hypercubes require least connections, with 3-D mesh and 3-D tori needing 2 and 3 times this number. Figure 5-17 illustrates the wide variation in the number of links required to construct these M-Ps.

Naturally, bus-based hypercubes have the lowest link requirements with less than 20 buses required for a 81-PE 2-D dual-bus hypercube. With only 1 bus in each orthogonal dimension, this scheme is $d / 2$-times cheaper than a spanning-bus implementation. Hypercubes constructed with point-to-point links require over 140 for a M-P of the same size.


Figure 5-17: Number of links for hypercube topologies.

### 5.4.3.2 Expansion Metrics

Since the number of nodes in a hypercube is $w^{d}$, all hypercube topologies of a given width $w$ and dimension $d$ have the same $n$ and, as a result, identical node expansion metrics. From a manufacturing perspective, lower dimension hypercubes have a definite advantage as lower node increments mean that 8 intact DMC system configurations exist with less than 81 -PEs.

Due to the wide variation in the number of links noted in Section 5.4.3.1, link increments also vary widely. By construction, 2-D spanning-bus and 2-D dual-bus hypercubes have a constant link increment of 2. Hypercube topologies based on point-to-point links expand very rapidly since in the order of $d$ links are required for each additional PE.

### 5.4.3.3 Network Connectivity

Link connectivities are constant and vary between 2 and 6 for the hypercube topologies considered here. The 2-D mesh, 2-D spanning-bus and all dual-bus hypercubes have a metric value of 2. Due to 6 - and 3-valent construction, the 3-D mesh and 3-D spanning-bus have a link connectivity of 3 . Incrementing $d$ gives a metric of 4 , the same as for the 2-D torus.

Under this metric, the best topology is the 3-D torus which can withstand 6 link failures before a PE is isolated. Figure 5-18 illustrates that the node connectivities for the point-to-point hypercubes (mesh and torus) are fixed by the node valency. Due to the construction of the bus-based hypercubes, node connectivity scales with $n$.


Figure 5-18: Node connectivities for hypercube topologies.

### 5.4.3.4 Diameter \& Valency

Bus-based hypercubes have a definite advantage over their point-to-point counterparts as $\delta \propto d$. Mesh diameter increases with $n$ giving $\delta=16$ for a 81 -node 2-D M-P. Comparable tori reduce this value by half as the topology is closed. Node valency for mesh and tori hypercubes is simply $2 d$, whereas dual-bus hypercubes only require a valency of 2 .

By construction, shared-bus hypercubes require PEs with $v=d$. From Figure 5-19, the $\delta \cdot v$ metric for hypercubes varies depending on the construction method employed. As the bus-based hypercubes have constant $v$ and a diameter proportional to $d, \delta \cdot v$ is constant. Consequently, both 2-D spanning-bus and 2-D dual-bus structures have a metric value of 4 .


Figure 5-19: $\delta \cdot v$ characteristics for hypercube topologies.

### 5.4.4 Preferred Topologies

Although bus-based hypercubes are attractive from a theoretical perspective, bus contention rapidly becomes a critical factor. Here, practical DMC M-P topologies are analysed together.

### 5.4.4.1 Expansion Metrics

Ideally, a DMC M-P should be incrementally expandible to afford greater design flexibility and a structured upgrade path. From Figure 5-20 it is clear that the scalability of many M-P architectures is restricted by topology-dependent factors if other characteristics are to be preserved. For example, the size of a binary hypercube can only be increased by doubling $n$.

Whereas shared-bus, chordal-ring and, to a lesser degree, 2-D hypercubes scale in modest increments, 3-ary trees and cube-connected cycles expand rapidly. Other factors are at work here: for example IPC can quickly saturate the nodes near the root of a simple $b$-ary tree as $n$ increases. Of course, extending a DMC is only important within commercial constraints.


Figure 5-20: Node increments for preferred M-P topologies.

### 5.4.4.2 Network Connectivity

Alternate paths ensure that the DMC cannot be partitioned by the failure of a single component. Unfortunately, technological constraints limit the number of connections per node. Table 5-2 shows that the tori and binary hypercube are most resistant to link failures though the chordal-ring, x-tree, 3-D mesh, and cube-connected cycles maintain a metric of 3 .


Figure 5-21: Node connectivities for preferred M-P topologies.

Link connectivity for the shared-bus is fixed at 1 as this topology is based on a single IPC channel, whereas Figure 5-21 indicates that the node connectivity is directly proportional
to $n$. Tori and the binary hypercube again perform well under this metric though, all things being equal, the most unreliable topology with regard to node failures is the simple $b$-ary tree.

### 5.4.4.3 Diameter \& Valency

Topology diameters vary over a range of 1 to 20 for the size of M-P architectures required for a DMC computation engine. Perhaps surprisingly, the chordal-ring is the worst topology with regards to this diameter since $\delta=20$, with the $2-\mathrm{D}$ mesh having $\delta=16$. All other prospective connectivity subsets, apart from the shared-bus, lie within the range 6 to 12 .

All topologies discussed in this section have a constant valency except for the binary hypercube where $v=d$. Figure 5-22 illustrates the $\delta \cdot v$ characteristics for these topologies. With the valencies required for mesh and chordal-ring configurations, these topologies are particularly expensive in terms of architectural costs incurred in order to minimise $\delta$.


Figure 5-22: $\delta \cdot v$ characteristics for preferred M-P topologies.

### 5.5 Conclusions

In this chapter, architectural issues pertinent to the construction of M-Ps for DMC use are developed. Equation (5-1) encapsulates a novel technique for estimating $n$, starting from the processing budget evaluated in the analysis of Section 3.2.1. The concept of processor pipelining and the fundamental real-time constraints on this scheme are established.

Thirteen single-stage M-P topologies are discussed and four of the more unfamiliar are analysed from first principles to determine their characteristics. A consistent framework of twelve metrics has been subsequently defined for these configurations. This schema has been used to construct the Topology Analysis Tool (TPG) deliverable described in Appendix C.

As demonstrated in Section 5.4, TPG allows the console designer to make a quantitative comparison of prospective DMC system architectures. In simple topologies, the number of links is linearly dependent on $n$. With a link expansion factor of one, the shared-bus is well suited to small-scale DMCs: however, contention quickly limits extensibility.

Ring architectures, particularly the tri-valent chordal ring, deserve attention from the DMC designer. Cross-topology links reduce the maximum inter-node distance $\delta$ to half that of the simple case. Point-to-point IPC channels avoid contention problems, while wormwhole routing, studied in Section 6.2.2, can minimise the effect of $\delta=20$ when $n=80$.

Expanding tree-based M-Ps by incrementing fanout $b$ cannot be recommended as the node valency is not fixed. Incrementing the number of levels $l$, however, fixes the number of IPC ports on each PE. Expansion factors ultimately converge to $b$, though the number of links required to extend an x -tree architecture is double that required for an equi-valent tree.

If valency is constant, PEs need not be changed to build larger DMCs. 2-D mesh, 2-D torus, and simple trees are the most attractive constant-valency M-Ps with small v. Less familiar topologies such as cube-connected cycles are also attractive for this reason. Despite being able to embed mesh and tree topologies, $v$ for a binary hypercube increases with $d$.

Although the simple $b$-ary tree appears attractive, it suffers greatly from an imbalance of inter-node traffic near the root node. Ring connections in the $x$-tree solve this problem while still corresponding closely to converging precedence constraints. Irrespective of these advantages, a PE with a valency of 5 is required to assemble a 2 -ary x -tree.

If a primary design goal is a wide product range, lower dimensional hypercubes and trees with restricted $b$ provide a structured upgrade path. While dual-bus hypercubes are extremely undesirable from a manufacturing perspective, hypercube topologies based on point-to-point links expand rapidly as $O(d)$ links are required for each additional PE.

In terms of reliability, the torus and binary hypercube are most resistant to link failures as $n c \propto d$ though at least 3 links must fail in the chordal-ring, x-tree, 3-D mesh and cubeconnected cycles before a processor is isolated. Unfortunately, the shared-bus is least reliable from this point of view as no IPC traffic is possible if the single IPC channel should fault.

Dense networks offer short inter-node distances but are more difficult to construct in practice as $n$ becomes large. However, altemative message-passing schemes can minimise the effect of large $\delta$. With this theoretical analysis complete, it is time to consider the question of how to assemble M-Ps suitable for DMCs with commercially available DSP technology.

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## Chapter 6

## Audio M-P Implementation

While the previous chapter investigates architectural issues from a theoretical view-point, this chapter focuses on the implementation of audio M-Ps using programmable DSPs. Messagepassing schemes are analysed and design notes presented for a hybrid M-P architecture.

### 6.1 Multi-DSP Configurations

A limited zero-chip interface may be implemented by connecting the serial ports of DSPs. For audio-rate IPC, the external memory interface (EMI) must be used to connect multiple DSPs.

### 6.1.1 Shared-Bus Design

Shared-bus architectures are probably the most common form of M-P, since they are the natural extension of conventional uni-processor DSP systems.

### 6.1.1.1 Introduction

The simplest way to construct an audio M-P is to connect DSPs on a shared-bus, as shown in Figure 6-1. In order to isolate DSPs so that they can simultaneously access local DSP RAM, bi-directional bus transceivers are required: the bus is then active only during IPC. Without local DSP RAM, bus contention would result in severe arbitration delay [Snell, 1989].

Devices such as the DSP56001 have no provision to force the EMI to quickly tri-state in a deterministic manner. Since the address bus is always driven, one DSP's local RAM cannot be addressed by another DSP while the first accesses internal memory. Additional bus transceivers stop the DSP56001 shorting its main bus, but increase board real-estate.

### 6.1.1.2 Arbitration Schemes

Arbitration schemes enable multiple processors to access the same piece of data at the same time. Most DSP devices have provision for asynchronous sharing of their EMI, with the use of


Figure 6-1: Example shared-memory M-P architecture.
bus-request ( $B R$ ) and bus-grant ( $B G$ ) signals. Architectures which use $B R$ and $B G$ signals directly typically require a hard-wired master-slave relationship between the DSPs.

There is little space in a 48 kHz sample period for BR/BG arbitration. In synchronous designs, the most serious problem is the indeterminate timing of BG. For example, the BR to BG delay for a 20.5 MHz DSP56001 is between 73 and 240 ns . Since the upper bound is greater than one instruction cycle, synchronous DSPs cannot be kept in lock-step.

### 6.1.1.3 Disadvantages of Bus Systems

An $n$-processor system requires an IPC media whose bandwidth is of the order of $n$-times that of a uni-processor. Consequently, the major shortcoming of shared-bus M-Ps is self-evident: IPC bandwidth is constant. As $n$ increases, the shared bus between DSP modules becomes a serious bottle-neck, fixing an upper limit on the extensibility of the design.

For limited $n$, bus-oriented systems can yield high performance/cost ratios without any pretensions of scalability. Beyond this range, delays due to contention and subsequent arbitration lead to considerable performance degradation. Such systems cannot support DMCs unless a technological breakthrough provides a very high bandwidth bus at relatively low cost.

### 6.1.2 Alternative Synchronization Mechanisms

Clock skew is a considerable problem in large synchronous M-P architectures. In asynchronous designs, synchronization mechanisms ensure that shared data is always valid.

### 6.1.2.1 Semaphores

Semaphore flags, first proposed by Dijkstra in 1965, are non-negative integer variables used as locks to shared resources [Dijkstra, 1965]. Two different operations ensure that only one PE gains access at once: a request operation which attempts to gain access and a release operation which signals access termination. Together these two processes guarantee mutual exclusion.

At one extreme a shared memory can be controlled by a single semaphore, whereas at the other every data item could be individually locked. The former precludes any parallel operations on shared data, while in the latter the overhead of locking operations is prohibitive. Typically, this overhead is distributed with one semaphore locking the corresponding data-set.

### 6.1.2.2 Test-and-Set

One popular semaphore technique is the test-and-set primitive. Firstly, the DSP requiring exclusive access reads the corresponding flag. It then tests this prior value to determine if it was successful (i.e. a ' 0 ') and, finally, forces the semaphore to ' 1 '. If the flag was not already set, then this DSP has permission to access the shared data: all other processors are locked out.

The test-and-set must be uninterruptible, otherwise synchronization could fail. One PE may test the semaphore and, before it can be rewritten, another PE may also initiate a test. Both processors then believe they have exclusive access. Most DSPs support an indivisible READ/MODIF Y/WRITE (RMW) instruction, ensuring that the address bus remains active.

### 6.1.2.3 Spin-Lock

Test-and-set is only one half of this mechanism: the other is the action taken depending on whether or not access has been granted. One technique which gives fast resource entry is spinlock. If the lock is granted, the PE continues on to use the shared resource. If not, then a spinlock occurs and the PE spins backwards to re-execute the test-and-set, as Figure 6-2.

There are, however, two drawbacks to this scheme. Firstly, a PE polling a semaphore does no useful ASP. Secondly, it consumes cycles in the memory containing the semaphore. If several DSPs are waiting at the same flag, contention rapidly consumes IPC bandwidth. Spinlocks can be wasteful, dedicating instruction cycles to repeatedly testing semaphores.


Figure 6-2: Flow chart of test-and-set and spin-lock semaphore operations.

### 6.2 Message-Passing IPC

Communication efficiency, one of the most important factors to be considered when designing a M-P architectures, often becomes one of the main obstacles to computation performance.

### 6.2.1 Overview

The simplest way to interconnect multiple DSPs for message-passing is to provide full connectivity. Another option is to employ some form of partial connection topology as discussed in the previous chapter. When a message passes between a pair of DSPs in a DMC, it may be routed through the M-P in a number of hops. Thus the communication speed of the architecture depends not only on the M-P topology but also the method of IPC used.

### 6.2.2 Routing Schemes

One method used to route messages between PEs requires intermediate PEs to store the entire message. Alternatively, the message may be directly transferred to the destination processor.

### 6.2.2.1 Store-and-Forward

Many first-generation M-Ps such as the Intel iPSC/1 binary hypercube use a store-andforward mechanism [Tanenbaum, 1981]. As messages pass indirectly between source and destination, each intermediate PE must temporarily store the message. While messages move between nodes, memory bandwidth and instruction cycles are consumed in intermediate PEs.

Communication latency is sensitive to the distance across the IN a message must be passed. As latency is linearly proportional to the number of hops, it can be expressed as:

$$
\begin{equation*}
T_{s}=T_{1} H \tag{6-1}
\end{equation*}
$$

where $T_{1}=K / B_{1}$ is the time for a message of $K$ bytes to pass through a link of bandwidth $B_{1}$ bytes/s. If $H$ is the number of hops, then $T_{1}$ is the routing delay of each node.

### 6.2.2.2 Wormhole

Wormhole routing is used in many second-generation M-Ps, like the Intel iPSC/2, to reduce message latency caused by topology diameter [Athas, 1988]. Instead of storing a complete message, only a few flow-control digits (flits) need be buffered at each node. Flits at the head of the message govern the route, while the remaining bits follow in pipeline fashion.

The communication latency of the wormhole routing scheme can be defined as:

$$
\begin{equation*}
T_{w}=T_{2} H+\frac{K}{B_{2}} \tag{6-2}
\end{equation*}
$$

where $T_{2}=k / B_{2}$ is the time required for the message head $k$ to pass through a channel of bandwidth $B_{2}$ bytes/s. $H$ is the number of hops as before, and $K / B_{2}$ is the time required to transmit the message of $K$ bytes continuously through the $B_{2}$ bytes/s wormhole channel.

### 6.2.2.3 Comparison

The ratio between Equations (6-1) and (6-2) is a quantitative comparison of the two schemes:

$$
\begin{equation*}
\frac{T_{s}}{T_{w}}=\left(\frac{B_{2}}{\overline{B_{1}}}\right) \frac{K H}{k H+K} \tag{6-3}
\end{equation*}
$$

Typically, $k « K$ and, as shown in Figure 6-3, Equation (6-3) can be approximated by:

$$
\begin{equation*}
\frac{T_{s}}{T_{w}} \approx\left(\frac{B_{2}}{B_{1}}\right) H \tag{6-4}
\end{equation*}
$$

Equation (6-4) indicates that wormhole routing reduces communication latency by up to $\left(B_{2} / B_{1}\right) \times H$ times over store-and-forward for similar message size $K$ and distance $H$.


Figure 6-3: Analysis of Equation (6-3), for various message head/size ratios.

For example, the first-generation iPSC/1 uses store-and-forward with a bandwidth of 1.25 Mbytes/s, whereas the iPSC/2 uses the wormhole routing and $4 \mathrm{Mbytes} / \mathrm{s}$. For a 32 -node hypercube $H=5$, indicating that a 32 -node $\mathrm{iPSC} / 2$ reduces communications latency by 16 times over a 32 -node $\operatorname{iPSC} / 1$. Even if $B_{1}=B_{2}, T_{w}$ is still reduced by a factor of $H$.

### 6.2.3 Comparing Message-Passing M-P

Intel iPSC systems are based on the binary hypercube, while many other commercial M-Ps use a 2-D mesh. In this section wormhole topologies are compared in terms of IPC latency.

### 6.2.3.1 Constant Bisection Bandwidth

As noted in Section 5.2.1.8, a $d$-D binary hypercube is created by duplicating the bisection, or ( $d-1$ )-D hypercube, and inter-connecting corresponding nodes from each. This approach
needs $n / 2$ links across the bisection: using the same method to construct a 2-D mesh requires $\sqrt{n}$ links. The diameter of a binary hypercube is $\log _{2} n$, and $\sqrt{n}$ for a 2-D mesh.

Assuming a constant bisection bandwidth, the 2-D mesh has $\sqrt{n}$ times more bandwidth per link than an $n$-node hypercube. For the maximum $H$, Equation (6-2) becomes:

$$
\begin{equation*}
T_{\mathrm{b}-\mathrm{hpc}}=T_{d} \log _{2} n+\frac{K}{\bar{B}} \tag{6-5}
\end{equation*}
$$

Since the link bandwidth in a 2-D mesh is $\sqrt{n}$ times wider than for a binary hypercube:

$$
\begin{gather*}
T_{\text {mesh }}=T_{d} \sqrt{n}+\frac{K}{B \sqrt{n}}  \tag{6-6}\\
\frac{T_{\mathrm{b}-\mathrm{hpc}}}{T_{\text {mesh }}}=\sqrt{n} \frac{T_{d} \log _{2} n+T_{c}}{T_{d} N+T_{c}} \tag{6-7}
\end{gather*}
$$

When $K$ is reasonably large, $T_{d}$ may be ignored, and the ratio in Equation (6-7) is $O(\sqrt{n})$.

### 6.2.3.2 Constant Node Bandwidth

In many M-Ps, the I/O pins of each DMC PE support a constant bandwidth $\beta$. Under this assumption, if the valency of a node is $u$, the bandwidth of each link is $\beta / v$. For a $d$-D binary hypercube $n=2^{d}$, the diameter $\delta=\log _{2} n$, and the valency $v=\log _{2} n$. From Equation (6-3), the maximum latency of binary hypercube using wormhole routing is:

$$
\begin{equation*}
T_{\mathrm{b}-\mathrm{hpc}}=T_{d} \log _{2} n+\frac{K \log _{2} n}{\beta} \tag{6-8}
\end{equation*}
$$

where $T_{d}=k \mathrm{v} / \beta$ is the delay of individual nodes on the wormhole path. Similarly, the metrics from Chapter 5 for a 2-D mesh, 2-ary x-tree, and chordal ring yield, respectively:

$$
\begin{gather*}
T_{\text {mesh }}=2 T_{d}(\sqrt{n}-1)+\frac{4 K}{\beta}  \tag{6-9}\\
T_{\mathrm{x}-\mathrm{tree}}=2 T_{d}(l-2)+\frac{5 K}{\beta}  \tag{6-10}\\
T_{\mathrm{c}-\mathrm{ing}}=T_{d}\left\lceil\frac{n}{4}\right\rceil+\frac{3 K}{\beta} \tag{6-11}
\end{gather*}
$$

The second term $K / B_{2}$ of Equation (6-2) dominates latency for all but very short messages. When $K » k, T_{d} \rightarrow 0$ giving the latency ratios in terms of valency as illustrated in Figure 6-4.


Figure 6-4: IPC latency ratios for several topologies given constant node valency.

### 6.3 Message-Passing Realisation

Various schemes have been suggested for message-passing between conventional DSP processors. Here, approaches which utilise widely available integrated devices are developed.

### 6.3.1 First-In First-Out Buffers

One common technique for providing an elastic data buffer between two asynchronous or asynchronous systems is the first-in first-out (FIFO) buffer.

### 6.3.1.1 Integrated FIFOs

Integrated products, such as the devices available from IDT, provide FIFO buffers at very fast speeds ( 50 ns ). Two self-incrementing pointers locate where reading (READ) and writing (WRITE) take place in a dual-ported circular-buffer. This approach results in very low fallthrough times ${ }^{1}$ compared to first-generation designs based on ganged shift-registers.

In second-generation designs, separate READ and WRITE ports allow for simultaneous asynchronous accesses without hand-shaking or arbitration. Additional logic provides empty

[^28](EF), half-full (HF), and full (FF) queue flags. Two additional pins, one input (XI) and one output (XO), provide for unlimited expansion while maintaining the 50 ns fall-through time.

### 6.3.1.2 Status Flags

Status flags allow the boundary conditions of integrated FIFOs to be monitored. As soon as a WRITE occurs, EF is de-asserted. When WRITE operations fill the buffer, FF flags that there are no more empty locations. If a READ reduces data to just below half-way, then HF is deactivated. EF and FF are also fed back internally, to eliminate data over- or under-flow.

Flagged FIFOs offer the features discussed above plus two additional programmable flags: ALMOST-EMPTY (AE) and ALMOST-FULL (AF). These flags can be used as early warning flags in real-time applications such as pipeline DSP. In a multi-tasking environment, these enhanced flags can set interrupt requests in advance to minimise IPC latency.

### 6.3.1.3 FIFO Memory Expansion

If the maximum FIFO requirement is 1024 locations of 9-bits, a single IDT7202 will suffice. Wider word widths can be achieved by connecting several devices in parallel. Status flags can be detected from any device because each device is working in lock-step parallel. Classical architectures require more external logic to account for differences in internal clocks.

In older architectures, fall-through time increased in direct proportion to the number of devices connected in series. A parallel architecture results with the two-pointer approach retaining the single-chip fall-through time. Since FIFOs do not have chip selects and external decoding mechanisms, XO and XI pins allow the pointer enable to cross device boundaries.

### 6.3.2 Dual-Ported RAM

In dual-ported RAM (DPR) two sets buses and control signals access one block of RAM. As a result, two DSPs can share the same physical memory in their respective address spaces.

### 6.3.2.1 Integrated DPR

Several manufacturers produce integrated DPR operating at static RAM speeds, shown in Table 6-2. Previously, few true DPRs were commercially available: designers configured external logic and single-ported RAM to simulate DPR operation. Integrated DPR devices
out-perform discrete parts designs as bus-access arbitration is no longer required. Nevertheless, synchronization must be performed at other levels to ensure data integrity.

| Width | Size | Part No. | Support Logic |  |  |  |
| :---: | :---: | :---: | :---: | :---: | :---: | :---: |
|  |  |  | Interrupt | Busy Logic |  | Semaphore |
|  |  |  |  | MASTER | SLAVE |  |
| $\times 8$ | 1K | IDT7130 |  |  |  |  |
|  |  | IDT7140 |  |  |  |  |
|  | 2 K | IDT7132 |  |  |  |  |
|  |  | IDT7142 |  |  |  |  |
|  |  | IDT71321 |  |  |  |  |
|  |  | IDT71421 |  |  |  |  |
|  |  | IDT71322 |  |  |  |  |
|  | 4 K | IDT71342 |  |  |  |  |
| $\times 16$ | 2K | IDT7133 |  |  |  |  |
|  |  | IDT7143 |  |  |  |  |

Table 6-1: Selection of dual-ported RAMs available from Integrated Device Technology.

### 6.3.2.2 Interrupt Logic

A common problem in DPR systems is signalling between processors. For example, DSP-1 needs to signal DSP-2 to request a task to be performed, and when DSP-2 has completed, it needs to signal DSP-1 that the task is done. Note that the signalling must occur in both directions. A common form of signalling is for one DSP to cause an interrupt on the other.

On certain DPR devices, the top two memory addresses also as dedicated interrupt generators. If DSP-1 writes into one of these addresses, an interrupt latch is set and the interrupt line to the other port activated. The interrupt latch is cleared when DSP-2 reads from this address. A similar set of logic is provided to allow DSP-2 to cause an interrupt on DSP-1.

### 6.3.2.3 Busy Logic

There are two significant cases: if one port is reading while the other is writing, data on the read side will change; if both ports attempt to write at the same time, the result can be random.

Busy logic solves this problem by detecting when both sides access the same location. The BUSY pins of DPR devices are suitable for attachment to the WAIT inputs of most DSPs.

Although one or other processor may have to wait occasionally, throughput loss is minimal. For example, in a 1 K word DPR with a relatively uniform and random access profile, the probability of any given location being accessed by one side is $O\left(10^{-3}\right)$. Average throughput can be reduced by only one part in a million due to DPR access contention.

### 6.3.2.4 Semaphore Support

Software constraints may require mutual exclusion of data structures rather than memory locations. Instead of comparing addresses on every cycle and occasionally asserting BUSY, other DPR devices support semaphores. Conventional test-and-set semaphores require that the two memory accesses are indivisible: some DPR devices employ a twist by using set-and-test.

The set corresponds to a request and the test checks to see if the request was granted. As the indivisible double access requirement is avoided, semaphores can be supported for DSPs which do not provide an indivisible RMW. Since there is no hardware relationship between semaphores and DPR locations, variable channel widths can be defined in software.

### 6.3.2.5 DPR Memory Expansion

Integrated DPRs can be combined to form larger dual-port memories. Expansion in depth is similar to that used for conventional RAM, where the address MSBs enable individual memory blocks. DPRs can also be expanded in width. If addresses for both ports arrive simultaneously it is possible for both sides to assert BUSY, causing both DSPs to live-lock.

Expanded DPRs used for DSP-to-DSP IPC is shown in Figure 6-5. Two DSP56001s communicate using three 8-bit wide DPR chips in a MASTER/SLAVE word-wide configuration. If the MASTER asserts BUSY, the BUSY input on the SLAVE devices disables their write enables. Note that the BUSY signals from the MASTER are also fed to the WAIT pins of the DSPs.

### 6.3.3 DPR and FIFOs Compared

A $2 \mathrm{~K} \times 24$-bit DPR block requires three 48 -pin ICs, plus address decode components. Using uni-directional devices, a bi-directional $2 \mathrm{~K} \times 24$-bit FIFO channel requires a total of $6 \times 28$ pin ICs. With each doubling in size, 2 address pins must be added to the DPR package. Larger


Figure 6-5: DSP-to-DSP communication using width-expanded dual-ported RAMs.
FIFOs use the same package size, since memory locations are not directly addressed. While FIFOs scales better than DPRs, the latter provide considerably more IPC flexibility.

### 6.3.4 Novel Devices

Recent technological developments in VLSI design have resulted in the emergence of novel integrated devices which have direct application in multi-DSP architectures.

### 6.3.4.1 Synchronous FIFOs

High-speed integrated FIFO may recognise noise-induced glitches as valid control pulses. Third-generation synchronous integrated FIFO designs reduce sensitivity to glitches via input and output registers. Independent external clocks force READ and WRITE operations to occur on clock edges. IDT's SyncFIFO products allow the depth of status flags to be programmed.

Many high-speed digital systems rely on a synchronous approach, since the complexity of asynchronous control circuitry increases with system clock frequency. In a synchronous system, the amount of control logic is minimal. If a design is made synchronous at the outset, later clocking improvements do not require redesign of control logic.

### 6.3.4.2 Quad-Ported RAM

Integrated quad-ported RAMs (QPRs) simplify the task of creating generalised multi-DSP systems for real-time ASP, all but preventing IPC contention. Although typically three times as expensive as comparable DPR devices, QPRs have a considerable advantage over DPRs. In order to fully interconnect four DSPs requires 1 bank of QPR as opposed to 6 banks of DPR.

As in DPR devices there are only two constraints on access patterns (see Section 6.3.2, Dual-Ported RAM). Hardware methods of controlling access - interrupts, semaphores and busy logic - and software-only token-passing schemes are possible. Separate locations dedicated to the status of each processor can also guarantee clean inter-task communications.

### 6.3.4.3 CalTech Torus Routing Chip

General purpose M-P routing components have been studied by several researchers. One such component is the Torus Routing Chip (TRC) developed at CalTech. This trivalent design was primarily designed to support efficient message-passing in 2-dimensional tori of width $w$. Two bi-directional links carry message traffic between distinct nodes: the third allows several chips to be cascaded permitting the construction of higher dimension tori.

### 6.4 HYMIPS - A Hybrid Audio M-P

Architectural issues studied in this and the previous chapter have led to the construction of a scalable hybrid M-P - HYMIPS - designed specifically for real-time, multi-channel ASP.

### 6.4.1 Design Decisions

In this section, design decisions surrounding the choice of routing and compute processors, interconnection components and memory maps for HYMIPS are considered.

### 6.4.1.1 Processors

The Inmos transputer is capable of parallel communication and computation, providing an ideal building block for scalable M-P architectures. IPC is accomplished by asynchronous point-to-point links which incur minimum CPU overhead [Inmos Ltd., 1989]. Unfortunately, as noted in Section 4.6.5, the transputer ALU is not efficient at executing ASP algorithms.

In contrast, DSPs such as the Motorola DSP56001 are specifically designed to implement such algorithms (see Section 4.6.1, Parallel-Bus DSPs). Although supporting a high degree of operational concurrency, M-Ps are not easily implemented with these devices alone. A hybrid M-P offers the required properties of scalability and efficient execution.

### 6.4.1.2 Interconnection

As discussed in Section 6.1.1, the fixed IPC bandwidth of the shared bus topology is a severe constraint on extensibility: bus contention results in substantial IPC latency. A variation of this architecture is to use DPR as the shared resource, as shown in Figure 6-6. Although the T801 and one DSP may simultaneously access DPR, DSPs still experience bus contention.


Figure 6-6: Dual-bus hybrid M-P architecture.

If this scheme is extended to provide each compute PE with a DPR channel to the transputer EMI as Figure 6-7, bus access arbitration is no longer required. IPC data may be mapped into the relevant DPR using the Occam PLACE instruction. A DSP processes data-set $n$ in parallel with the routing processor transferring $n-1$ and $n+1$ via DPR channels.

### 6.4.1.3 Memory Maps

While the DSP56001 has 3 independent memory spaces, only one can be accessed via the EMI in any cycle. 2 Kwords of DPR is mapped to a sufficiently high base address to allow


Figure 6-7: Multi-bus hybrid M-P architecture.
modulo-addressing across the whole DPR. ${ }^{1}$ An address-decode PAL allows half of the DPR to be mapped into P-space, so that the T801 can place object code directly into P.

The T801 can access a single address space of 1 Gwords, with 1 Kword placed onchip. 8 Kwords of local memory is contiguous with the on-chip RAM, allowing internal program and workspace areas to overflow into external memory. Alternative DPR memorymaps may be implemented by replacing the transputer address-decode daughter-board.

### 6.4.2 Message-Passing

Although the T801 itself implements store-and-forward IPC, the compute PEs in Figure 6-7 can utilise a pseudo-wormhole scheme as they do not handle messages for other PEs.

### 6.4.2.1 Semaphore Protocol

As the transputer is designed to perform IPC over serial links in a CSP context, the EMI does not support an indivisible RWM. Consequently, a modified semaphore scheme is required to avoid the problems discussed in Section 6.1.1. Previously, the semaphore indicated if the associated channel is locked by one of, possibly, several processors.

In HYMIPS, DPR channels are shared between two processors allowing a different semaphore protocol to be implemented. Rather than flagging whether or not the channel is

[^29]locked, the token determines which processor has exclusive access rights. Together with the DPR on-chip arbitration logic, this modified protocol ensures data-set integrity.

### 6.4.2 2 Inter-Node Communication

Inter-node communication is achieved using transputer links. With a bi-directional bandwidth of $2.35 \mathrm{Mbytes} / \mathrm{s}$, each link supports 12 digital audio channels. One link can be reserved for broadcasting control data to the DSPs via a IMSC011 link adaptor [Inmos Ltd., 1989]. In this configuration each transputer can support up to 36 inter-node audio channels.

Intra-node communication is achieved through the non-multiplexed EMI of the T801. Each DSP possesses an exclusive block of DPR: the other port of each DPR is mapped into the transputer's EMI. As a 25 MHz T801 transputer can transfer data over its non-multiplexed EMI at a rate of 12.5 Mwords/s, providing 130 digital audio paths within the $H Y M I P S$ node.

### 6.4.2.3 Intra-Node Reconfiguration

Although the physical topology of the HYMIPS node is fixed the logical topology is not, however. As the T801 code determines intra-node IPC, the organisation of compute PEs may be altered dynamically to give a variety of virtual topologies. An aliased memory-map is included in the transputer address-decode PAL to enhance broadcast IPC performance.

One DSP56001 memory mapping mode addresses half of the DPR as P-space, allowing the T801 to down-load ASP tasks to the DSP. Object kernels may be stored in local transputer memory or read from the host file-system via a link. For efficiency reasons, the DSP must then move the object code into local DSP RAM or, preferably, internal P-space.

### 6.4.3 Performance

The computation performance of $H Y M I P S$ is limited by intra- and inter-node IPC bandwidth. Two message-passing schemes were proposed: orthogonal and pipelined. Using an orthogonal IPC hamess, IPC channels are transferred over transputer links. As the T801 has four bidirectional links, this limits the number of.DSPs that may be serviced in any IPC cycle.

In the pipelined harness, only one I/P channel and one $O / P$ channel are serviced directly by hardware links: all other IPC is intra-node. With 4 bi-directional links and an IPC
channel of 256 words, Gould has shown that HYMIPS node can support 17 DSP56001s per node with zero IPC latency, giving a total node performance of 175 MIPS [Gould, 1992].

### 6.4.4 SP-DIF Interfaces

Motorola's ADS56016 EVB demonstrates the ease with which data-domain converters can be assembled. To use HYMIPS in an all-digital audio path requires direct digital audio I/O.

### 6.4.4.1 Motorola DSP56001 PORT C

The nine I/O pins (PC0-PC8) of the DSP56001 PORT C support serial communications (SCI) and synchronous serial (SSI) interfaces. PC0-PC2 can be configured as specific SCI pins or for general-purpose I/O. To interface with Motorola SPI peripherals, the SSI has three pins dedicated to transmit data (STD), receive data (SRD) and a bi-directional serial clock (SCK).

Three other pins - SC0, SC1 and SC2 - may also be used to control bits in the SSI status register. In synchronous clock modes, SC0 and SC1 can addresses multiple peripheral devices. SC2 supports frame sync for both transmitter and receiver [Motorola Corp., 1989]. In HYMIPS, SC0 and SC2 have been mapped onto LR and WC as shown in Figure 6-6.


Figure 6-8: Timing diagram for HYMIPS SP-DIF interfaces.

### 6.4.4.2 SSI Programming Model

From a coding perspective, the SSI can be viewed as two 16 -bit control registers (CRA, CRB), one read-only status register (SSISR), a read-only receive register (RX), and a write-only transmit register (TX). CRA controls the clock generator and frame sync rates and word length. CRB controls the multifunction pins, interrupt enables, and SSI mode (see Figure 6-9).

SSISR is an 8-bit read-only status register used to interrogate the SSI status and serial input flags. RX register accepts 24 -bit data from SRD when the receive shift register becomes

```
; Initialisation macro for HSR/HST SP-DIF peripherals
SPDIF_INI \
    MACRO
    MSG 'spdif_ini...'
    ; clear bus control register, zero wait-states
    movep #$000000,x:BCR
    ; set SSI RX interrupt priority level
    movep #$003000,x:IPR
    ; set continuous external clock, 16 bits word-length, word frame sync
    movep #$004000,x:SSI_CRA
    ; enable SSI RX interrupt, RX and TX, synchronous mode
    movep #$00B200,x:SSI_CRB
    ; enable port C pins PC3-PC8 as SSI
    movep #$0001FF,x:PCC
    MSG 'spdif_ini OK'
    ENDM
```

Figure 6-9: Initialisation macro for HYMIPS SP-DIF peripherals.
full. If the associated interrupt is enabled, the DSP56001 CPU is interrupted as Figure 6-7. Data written to TX is automatically transferred to STD via the transmit shift register.

```
; -----------------------------------------------------------------------------
; Interrupt service for HSR/HST SP-DIF peripherals: normally mapped to p:$000C
SPDIF_ISR \
    MACRO lin,rin,acc,lout,rout
    MSG 'spdif_isr...'
    ; move sample from SSI receive register
    movep x:SSI_RX,acc
    ; test SCO for left/right channel
    btst #0,x:SSI_SR
    .IF <CS>
                ; transfer left channel
            move acc,x:lin
            movep x:lout,x:SSI_TX
    .ELSE
            ; transfer right channel
            move acc,y:rin
            movep y:rout,x:SSI_TX
    .ENDI
    ; return from interrupt service
    rti
    MSG 'spdif_isr OK'
    ENDM
```

Figure 6-10: Interrupt service routine for HYMIPS SP-DIF peripherals.

### 6.4.4.3 HYMIPS SP-DIF Receiver

The YM3623B, developed by Nippon Gakki, was used in conjunction with TTL logic to implement a SP-DIF receiver (HSR) for HYMIPS. In the absence of optic fibre or coaxial SP-DIF, the internal PPL self-oscillates using the external crystal oscillator. LR indicates to which channel the current sample belongs, in sync with the MSB-first audio data on SRD.

If a parity error is detected in this design, audio data preceding the error is re-output. Copy disabled (DIS/CPY), error status (ERR), de-emphasis (DEF), source (CD/DAT) and sampling frequency are displayed on front-panel LEDs. As illustrated in Figure 6-8, the error status is also output on SC1. Table 6-4 lists the components identified in the HSR schematic.

| Component Type | Schematic ID | Value |
| :--- | :---: | :---: |
| capacitor | C 1 | 4.7 nF |
|  | $\mathrm{C} 2-\mathrm{C} 3$ | 10 pF |
|  | C 4 | 220 nF |
|  | C 5 | $47 \mu \mathrm{~F}$ |
| diode | C 6 | $100 \mu \mathrm{~F}$ |
|  | D 1 | IN 4140 |
|  | L 1 | $101 \mu \mathrm{H}$ |
|  | $\mathrm{R} 1-\mathrm{R} 2$ | $4.7 \mathrm{k} \Omega$ |
|  | R 3 | $150 \Omega$ |
|  | R 4 | $1 \mathrm{M} \Omega$ |
|  | R 5 | $1 \mathrm{k} \Omega$ |
| crystal oscillator | $\mathrm{R} 7-\mathrm{R} 14$ | $470 \Omega$ |
| zenner diode | R 15 | $21 \mathrm{k} \Omega$ |

Table 6-2: Component list for HSR schematic.

### 6.4.4.4 HYMIPS SP-DIF Transmitter

As shown in the schematic of Figure 6-9, the HST board is based on the Yamaha YM3613B. A built-in PLL synchronises with the sampling frequency of the digital audio input to this

Figure 6-11: Schematic diagram of HSR SP-DIF peripheral.

24-pin CMOS DIP. Internal timing siguals are generated by an external 16.93 MHz crystal connected across pins 3 (XI) and 4 (XO). Serial audio data on STD is assumed to be MSB first.

Although this design permits ' $P$-W' sequence data to be inserted, the overhead of subcode information was discarded in HYMIPS. Both optic fibre and coaxial outputs are driven at the same time, the latter via a VC54820 line transformer. A front panel LED indicates the presence of data errors. Table 6-4 lists the values of components identified in Figure 6-9.

| Component Type | Schematic D | Value |
| :--- | :---: | :---: |
| capacitor | $\mathrm{C} 1-\mathrm{C} 2$ | 10 pF |
|  | C 3 | $100 \mu \mathrm{~F}$ |
|  | C 4 | $10 \mu \mathrm{~F}$ |
|  | C 5 | 220 pF |
|  | C 6 | 100 nF |
| resistor | C 7 | 10 nF |
|  | C 8 | $47 \mu \mathrm{~F}$ |
|  | R 1 | $1 \mathrm{M} \Omega$ |
|  | R 2 | $470 \Omega$ |
| crystal oscillator | R 3 | $2.2 \mathrm{k} \Omega$ |
| Zenner diode | R 4 | $75 \Omega$ |

Table 6-3: Component list for HST schematic.

### 6.5 Conclusions

Equations (6-1) and (6-2) show that IPC latency for message-passing schemes is dependent on the channel bandwidth, distance and message size. Compared to store-and-forward, wormhole routing greatly reduces IPC latency as delay is no longer sensitive to the distance involved in message transport. From Equation (6-7), the IPC latency in a 2-D mesh may be reduced up to $O(\sqrt{n})$ times over a binary hypercube if constant bisection bandwidth can be maintained.

Figure 6-12: Schematic diagram of HST SP-DIF peripheral.

Fewer links contribute to the bisection, which permits each channel to be made wider. Throughput is bounded, however, by requiring more channels to cross the bisection for higher dimensions. Figure $6-4$ reveals that there is a $21.26 \%$ advantage in reconfiguring an $80-$ processor binary hypercube as a 2-ary x-tree when node bandwidth is constant. For chordalring and 2-D mesh latency is reduced by $53.67 \%$ and $37.41 \%$ respectively for constant $n$.

Assuming constant PE bandwidth, IPC latency can be reduced by a up to a factor of 2 if a higher-dimensional $n$-processor hypercube is reconfigured as a 2-D mesh. X-tree topologies do not perform especially well under these conditions since 2 connections are required per processor to interconnect the ring at each level. A simple 3-ary tree gives the same wormhole latency improvement over a binary hypercube as a chordal-ring when $k$ « $K$.

Each HYMIPS PE occupies its own PCB with boards connected via a backplane bus. An IMSC011 link adaptor is provided on the T801 card to drive the DSP HOST ports with control parameters via link 0 . All links and the C011 are accessible via the front panel. DSP56001 boards incorporate a self-overwriting boot EPROM, DPR and a RS232/TTL converter for host access to the SCI. SSI and HOST ports are also available on the front panel.

As this is not a shared-bus M-P in the classical sense, performance is not limited by bus contention. Shared DPR and a modified semaphore protocol allow pseudo-wormhole messaging to overlap with computation. A HYMIPS node has been constructed with 2 compute PEs, together with HSR and HST SP-DIF peripherals. Independent testing confirms that computation performance aligns closely with theoretical predictions in Section 6.4.3. ${ }^{1}$

As the comprehensive design notes intimate, HYMIPS offers scalability both in terms of the number of compute PEs per node and the number of nodes. Topologies which require a node valency greater than 4 are possible with the addition of hardware switches such as the IMSC004 link crossbar. For further information, the interested reader should consult the paper presented by the author at ICASSP ${ }^{2}$ and included in this thesis as Appendix F [Linton, 1991].

[^30]
### 6.6 References

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## Chapter 7

## Multi-Processor Models

On M-P architectures like HYMIPS, load balancing tries to distribute tasks evenly whereas minimising IPC tends to assign the whole taskforce to one PE. As a conflict exists between these two aims, a compromise must be made to obtain good performance for a given DMC.

### 7.1 DMC Task Scheduling

A critical aspect of DMC design, on which the third part of this thesis focuses, is the development of algorithms for scheduling partially ordered tasks on multi-DSP topologies.

### 7.1.1 Introduction

The basic digital console scheduling (DCS) model consists of a taskforce of $m$ tasks, $T=\left\{T_{1}, \ldots, T_{m}\right\}$, and a M-P of $n$ identical PEs, $P=\left\{P_{1}, \ldots, P_{n}\right\}$. With each task $T_{i}$ is associated the number of instruction cycles to execute it. A scheduling algorithm allocates $T$ onto $P$, ordering the execution of tasks. No PE can execute more than one task at any time.

A DMC can be represented by a directed acyclic graph (DAG) where each open circle represents a processing task $T_{i}$ and each arc a precedence constraint. ${ }^{1}$ As shown in Figure 7-1, DCS can be visualised as a graph-matching exercise. If $m>n$, at least two tasks will be assigned to the same PE. However, multiple tasks may be co-resident even if $m<n$.

### 7.1.2 Scheduling Algorithms

A scheduling algorithm is said to solve a problem if it can be applied to any instance and is guaranteed to always produce a feasible solution. A feasible schedule is an assignment which does not violate any resource, precedence or preemption constraint. The degree of optimality, or cost, may be expressed formally in terms of an objective function [ $\mathrm{Hu}, 1961$ ].

[^31]

Figure 7-1: Illustration of the DMC task scheduling problem.
Given that a DMC taskforce is periodic, processing real-time sample trains, schedule length and accumulated latency are chosen as objective functions. In general, the most 'efficient' algorithm is required. In the context of DCS, the notion of efficiency primarily involves execution time as this factor determines if an algorithm is useful in practice.

### 7.1.3 Algorithm Constraints

By inspecting Figure 7-1, several pertinent observations can be made. Most notably, it is apparent that a scheduling algorithm is restricted by diverse technological constraints.

### 7.1.3.1 Precedence

Individual tasks within a taskforce can be related to each other in a number of different ways. In some instances, it is possible for all tasks to be independent; in others, ordering restrictions may be represented by a tree. Assuming that tasks can be executed in arbitrary order is not justified for DCS. Audio mixing requires a general precedence structure - 'a forest of trees'.

In DMCs, one task must be completed before another can begin execution. This type of constraint is incorporated into the DCS model as follows. Directed arcs between tasks in Figure 7-1 imply that a partial ordering, or precedence relation <, exists between tasks. Thus, if $T_{i}<T_{j}$, the execution of task $T_{i}$ must be completed before $T_{j}$ can be initiated.

### 7.1.3.2 Task Preemption

In non-preemptive schedules, a processor assigned to a task is dedicated to that task until it is completed. In general, preemptive disciplines generate better schedules than those generated by non-preemptive strategies. ${ }^{1}$ Nevertheless, certain penalties such as task switching overhead are incurred in preemptive schemes. As, by construction, atomic ASP tasks cannot be further partitioned, preemptive schedules are not considered further in this thesis.

### 7.1.3.3 Cycles and Conditional Tasks

The taskforce shown in Figure 7-1 is acyclic. A loop would prevent static scheduling, as the loop condition cannot be resolved until run-time. Most published work on scheduling ignore the difficulties presented by loops. However, cycles can be removed by either loop unrolling or by bisecting cycles across adjacent sample periods as depicted in Figure 7-2.

Also, the taskforce of Figure 7-1 contains no decision tasks. A decision task is a datadependent branch whose run-time outcome can affect the flow of control. While Lee (et al.) have developed techniques to deal with conditionals in general DSP, the requirement for such constructs is of minimal importance in real-time DMC computation engines [Lee, 1987].

### 7.1.3.4 Processor Idleness \& Processing Lag

Performance metrics can often be improved by deliberately idling a PE, at the expense of computational complexity. In fact, enumerative algorithms are only guaranteed to be optimal

[^32]

Figure 7-2: Illustration of loop removal by sample period bisection.
if the option of idling is permitted instead of scheduling a ready task. In 'greedy' DCS, no idle periods are inserted and a pending task is begun as soon as a PE is available.

Most important for real-time ASP applications, the DCS algorithm must ensure that accumulated processing lag - the result of processor pipelining - is not apparent. Experience indicates that all input-to-output delays through the DMC should be less than 1 ms (see Section 4.2.3) to facilitate real-time procedures such as overdubbing and track-bouncing.

### 7.1.4 Static vs. Dynamic Scheduling

There are two primary approaches to parallel processing and task scheduling on a M-P architecture. In the static approach, all parallel tasks in the problem taskforce are known before-hand. Consequently the taskforce can be statically mapped onto the processors at compile-time and remain fixed throughout run-time. This paradigm corresponds naturally to an DMC software design consisting of a network of communicating tasks.

In the dynamic approach, ASP computation is defined by a dynamically created taskforce and allocation decisions must be postponed until run-time, incurring substantial overheads. In DMC systems, the set of DSP algorithms and their resource requirements are known in advance. Consequently, the taskforce can be efficiently distributed beforehand, and the overheads associated with dynamic allocation eliminated.

### 7.1.5 Model Operation

With the foundation laid for DCS in terms of constraints and relationship to compile- and runtime, it is appropriate to consider how a DCS algorithm may fulfil the allocation role.

### 7.1.5.1 Input Processing

The operation of the DCS model begins at time $t=0$ with all $n$ PEs scanning the taskforce in search of ready tasks. A task is said to be ready for execution if all its predecessors have been completed and no processor has begun to execute it. If two or more processors compete for a task $T_{i} ; 1 \leq i \leq m, T_{i}$ is assigned to the processor $P_{j} ; 1 \leq j \leq n$ with the lowest index $j$.

When a processor cannot find a ready task, it stops scanning and becomes idle until a task is completed. As this task might be a predecessor of other tasks, whenever a task is completed, all idle processors instantaneously begin to scan the taskforce. The schedule ends when the final task is completed: following convention, the finishing time is denoted by $\omega$.

### 7.1.5.2 Output Processing

The run-time of the scheduling algorithm employed provides a second important performance criteria. In order to fully compare schedules obtained for various problem instances, a further performance metric is defined. The efficiency of a processor $P_{j}$ can be defined as total active processing time of $P_{j}$ divided by the schedule finishing time $\omega$ [Coffman, 1976].

Gantt charts are used to represent the timing of DMC taskforce schedules. Each task is represented by a time-space span whose length corresponds to its execution time. A horizontal row of time-space spans represents the sequence of tasks executed by a processor. Crosshatched regions indicate periods during which the corresponding processor is idle.

### 7.1.6 Extensions

In this section, some extensions to the basic constraints discussed above are considered. Both additional resource constraints and heterogeneous M-Ps are directly applicable to DCS.

### 7.1.6.1 Additional Resource Constraints

Most effort to date has assumed the unlimited availability of secondary resources. ASP tasks require more resources than just a compute PE, however. Most references assume that
sufficient memory is available to contain the program and data of assigned tasks. Of course memory is not the only secondary resource: another is the availability of SP-DIF I/O ports.

As might be expected, worst-case behaviour degrades further as the number of resources is increased. Interestingly, CPM still guarantees the same worse-case bound for one additional resource and $m>n$ [Garey, 1978]. ${ }^{1}$ A related observation is that investing further effort in the preparation of priority lists can lead to better results [Yao, 1974].

### 7.1.6.2 Heterogeneous Multi-Processors

Up till now, it has been assumed that all processors are identical. In heterogeneous M-P systems, a DSP can be replaced with a higher clock rate part or even a different processor altogether. This enhanced model has been studied from a theoretical performance-guarantee viewpoint by Gonzalez [Gonzalez, 1978]. The case where task times may vary arbitrarily from processor to processor has been treated, for independent tasks, by Bruno [Bruno, 1974].

### 7.1.6.3 Deadline-Driven Schedules

Manacher has considered the case in which terminal and non-terminal tasks require different completion times [Manacher, 1967]. Tasks with unequal execution times and precedence constraints are executed in a non-preemptive manner. Manacher's heuristic solution is a variation of the longest-path schedules considered in the next chapter. Multiple critical paths are defined for those tasks contained in paths that contain tasks with deadlines.

### 7.1.6.4 Job Periodicity

The majority of investigations deal with only a single execution of taskforce. Although the analysis required to generate near-optimal schedules can be significant, this is justified by the time saved during each of these executions. Consideration has also been given to the use of M-P in a control environment. Kafura has characterised an environment of this type by a set of tasks each of which has a known periodicity and processing time [Kafura, 1974].

[^33]
### 7.2 Modelling M-P Speed-Up

A model of $m$ tasks executing on $n$ PEs is presented which is capable of explaining various speed-up characteristics. The model is then considered in the context of DMC and DCS.

### 7.2.1 Introduction

'Superlinear' speed-up has been used to describe a computation using an $n$-PE M-P which is more than $n$-times faster than the uni-processor (U-P) case. Often this term is inappropriate as $S(n)$ may be linear and yet still greater than $n$. In this thesis, the terms superunitary, unitary, and subunitary speed-up are used when $S(n)>n, S(n)=n$, and $S(n)<n$, respectively.

Superunitary (SU) speed-up is only possible when the total amount of work performed by $n$ processors is strictly less than that performed by 1 . Hypothesised upper bounds on $S(n)$ range from Minsky's rather pessimistic $O(\log n)$ to unbounded [Minsky, 1970]. Conversely, there have been reports of SU speed-up in real applications [Sanuinetti, 1986].

### 7.2.2 Preliminary Conditions

The total number of instructions executed when a taskforce is processed can be represented by $w$. Each $w_{i}$ then indicates the number of times tasks of type $i$ are executed. To determine $\omega$, $r_{i}(n)$ weights the time required to execute one type- $i$ task on a homogeneous $n$-PE M-P. If one PE executes while the other $n-1$ are idle in sequential mode, then $r_{s}(n)=n r_{s}(1)$.

For notational convenience, $r_{s}(1)=1$. Separating the DMC taskforce into tasks which must execute sequentially and those with at least $n$-fold parallelism gives:

$$
\begin{equation*}
w=w_{s}+\sum_{i=1}^{m^{\prime}} w_{i} \tag{7-1}
\end{equation*}
$$

where $m^{\prime}$ is the number of tasks that exhibit $n$-fold parallelism and $w_{s}$ is the number of operations that must be performed sequentially. Then, $S(n)>n$ if and only if:

$$
\sum_{i=1}^{m^{\prime}} w_{i}\left(r_{i}(1)-r_{i}(n)\right)>(n-1) w_{s} r_{s}(1)
$$

### 7.2.3 Opportunities for SU Speed-Up

In this section, the model of Section 7.3.2 is used to investigate previously reported occurrences of SU speed-up behaviour in a variety of M-P environments.

### 7.2.3.1 Conventional Speed-Up

Equation (7-7) clearly states the conditions necessary for SU speed-up. If $w_{p}$ and $c_{p}(n)$ represent the amount and run-time cost of work that exhibits $n$-fold parallelism then the argument against SU speed-up characteristics can be stated as follows:

$$
\begin{gather*}
w=w_{s}+w_{p}  \tag{7-3}\\
r_{s}(n)=n r_{s}(1)=n  \tag{7-4}\\
r_{p}(n)=1 \tag{7-5}
\end{gather*}
$$

Equation (7-7) becomes $w_{p}\left(r_{p}(1)-r_{p}(n)\right)>(n-1) w_{s}$, and since the left-hand side is 0 , SU speed-up is impossible. In fact, if $w_{s}$ is positive, then the speed-up becomes:

$$
\begin{equation*}
S(n)=\frac{w_{s}+w_{p}}{w_{s}+\frac{w_{p}}{n}} \leq 1+\frac{w_{p}}{w_{s}} \tag{7-6}
\end{equation*}
$$

and Amdahl's law is obtained, as illustrated in Equation (7-6) [Amdahl, 1967].


Figure 7-3: Illustration of conventional speed-up for various values of $w_{p}$.

### 7.2.3.2 Increasing Cache Size

SU speed-up can occur when each. PE contains a local program or data cache. In this case, not only does the number of DSP ALU cores increase with $n$, but so does the total amount of cache memory. Whether or not a processor's local cache should be considered part of the processor depends on what is being measured. However, it is unreasonable to suppose that the cache size of a single DMC compute PE can be increased to match that of the equivalent M-P.

### 7.2.3.3 Hiding Latency

From Section 6.4, several methods can be used to hide IPC latency. Excess process-level parallelism can be achieved via time-slicing, reducing idle time at the expense of additional context-switching overhead. Alternatively, if the PE has DMA IPC capability, latency is concealed by utilising additional parallelism beyond that explicitly described. Since both schemes can be used on a U-P, they are optimisations rather than sources of true SU speed-up.

### 7.2.4 Scaled Speed-Up

In the previous sections, a given taskforce was executed on an extensible M-P architecture. A revised speed-up metric reflects cases where both $n$ and $m$ are increased [Helmbold, 1990].

### 7.2.4.1 Overview

Let $w_{s}(m, n)$ and $w_{p}(m, n)$ represent the number of sequential and parallel operations required to process a taskforce of size $m$ on $n$-processors. If $r_{s}(n)$ and $r_{p}(n)$ are defined as before, the time taken to process a taskforce of size $m$ on $n$ PEs, $\omega(m, n)$, is then:

$$
\begin{equation*}
\omega(m, n)=\frac{w_{s}(m, n) r_{s}(n)+w_{p}(m, n) r_{p}(n)}{n} \tag{7-7}
\end{equation*}
$$

A reasonable assumption is that the number of instruction cycles required by the taskforce is independent of the number of processors, $n$. Therefore, $w_{s}(m, 1)=w_{s}(m, n)$ and $w_{p}(m, 1)=w_{p}(m, n)$, which can be abbreviated to $w_{s}(m)$ and $w_{p}(m)$, respectively.

### 7.2.4.2 Definition

Following convention, the scaled speed-up (SSU) on an $n$-PE system $S^{\prime}(n)$ is defined as the ratio of the execution time of a taskforce of size $n$ on a U-P to that on an $n$-processor M-P:

$$
\begin{equation*}
S^{\prime}(n)=\frac{\omega(n, 1)}{\omega(n, n)}=\frac{w_{s}(n, 1) r_{s}(n)+w_{p}(n, 1) r_{p}(n)}{\frac{1}{n}\left(w_{s}(n, n) r_{s}(n)+w_{p}(n, n) r_{p}(n)\right)} \tag{7-8}
\end{equation*}
$$

Assuming $c_{p}(n)=1$ and $c_{s}(n)=n$ as before, Equation (7-13) simplifies to:

$$
\begin{equation*}
S^{\prime}(n)=\frac{n\left(w_{s}(n)+w_{p}(n)\right)}{n w_{s}(n)+w_{p}(n)} \tag{7-9}
\end{equation*}
$$

Since $w_{s}(n)+w_{p}(n)$ is always less than $n w_{s}(n)+w_{p}(n)$, provided $w_{s}(n) \neq 0$, unitary SSU is impossible. However, unitary SSU does occur in the limit when:

$$
\begin{equation*}
\frac{w_{s}(n)+w_{p}(n)}{n w_{s}(n)+w_{p}(n)} \geq \varepsilon \tag{7-10}
\end{equation*}
$$

for some positive constant $\varepsilon$. From above, the following must hold for linear subunitary SSU:

$$
\begin{equation*}
\frac{w_{p}(n)}{w_{s}(n)} \geq \frac{\varepsilon n-1}{1-\varepsilon} \tag{7-11}
\end{equation*}
$$

### 7.2.4.3 Problem Size \& Execution Speed

One reason for considering scaled speed-up is the question of how large a problem can be solved within some fixed time, rather than how quickly a given problem can be solved. It is reasonable to assume that the total work grows no faster than $O(n)$, i.e. $w_{s}(n)+w_{p}(n) \leq \alpha n$. Under this assumption, $w_{s}(n)$ must be bounded by a constant to as shown below:

$$
\begin{equation*}
w_{s}(n) \leq \alpha n-w_{p}(n)=\alpha n-w_{s}(n) \frac{\varepsilon n-1}{1-\varepsilon}=\frac{(1-\varepsilon) \alpha n}{\varepsilon(n-1)} \tag{7-12}
\end{equation*}
$$

This analysis contrasts sharply with the corresponding non-scaled case, where $S(n)$ is always bounded by a constant. Once $w_{p}(n)$ and $w_{s}(n)$ are known, the problem size $m$ can be fixed for (unscaled) speed-up and curves plotted for Equation (7-6) and Equation (7-9). As illustrated in Figure 7-4, the curves intersect where $S(n)=S^{\prime}(n)$ or:

$$
\begin{equation*}
\frac{w_{p}}{w_{s}}=\frac{w_{p}(n)}{w_{s}(n)} \tag{7-13}
\end{equation*}
$$

### 7.3 Linear Processor Pipeline

Here, the linear processor pipeline (LPP) of Section 5.1.4 is re-examined in detail. Several performance metrics are defined in terms of this topology, which supports a linear DAG.


Figure 7-4: Illustration of conventional and scaled speed-up curves. ${ }^{\text {a }}$
a. For the conventional speed-up curve, $S, w_{p}=98$ and $w_{s}=2$. In scaled curves, $S, S^{\prime \prime}$, and $S^{\prime \prime}$, $w_{s}(n)=\sqrt{n} / 5, w_{s}(n)=2$, and $w_{s}(n)=n^{2} /(n+4900)$, respectively, with $w_{p}(n)=n-w_{s}(n)$ in each case.

### 7.3.1 Background

The sample period $\tau$ of a homogeneous processor pipeline is the reciprocal of the sampling frequency, i.e. $\tau=1 / f_{s}$. As shown in Figure 5-4, a space-time diagram illustrates the overlapped tasks in an LPP M-P architecture. Once the pipeline is full, maximum performance gives one output for each sample period independent of the number of stages $k$.

Ideally, an LPP with $k$ stages can process $n$ audio samples in $T_{k}$ sample periods:

$$
\begin{equation*}
T_{k}=k+(n-1) \tag{7-14}
\end{equation*}
$$

given that $k$ sample periods are required to fill the pipeline and process the first sample, and $n-1$ to complete the remaining $n-1$ samples. Processing the same number of samples on a non-pipelined architecture with identical $f_{s}$ gives the number of sample periods $T_{1}$ as:

$$
\begin{equation*}
T_{1}=n k \tag{7-15}
\end{equation*}
$$

### 7.3.2 Speed-up

Consequently, the speed-up of a $k$-stage LPP over an equivalent non-pipelined M-P is:

$$
\begin{equation*}
S(k)=\frac{T_{1}}{T_{k}}=\frac{n k}{k+(n-1)} \tag{7-16}
\end{equation*}
$$

It should be noted that unitary speed-up, $S(k) \rightarrow k$, can be achieved by a processor pipeline if $n » k$, as illustrated in Figure 7-3. This maximum is never fully achieved due to data dependencies between tasks, interrupt handlers, control branches and other factors.


Figure 7-5: Speed-up characteristics for LPP M-Ps.

After describing speed-up in Equation (7-16), two related performance measures can be developed. First, the product of a time interval and a PE space in the space-time diagram of Figure 5-4 is defined as a time-space span. From the constraints of Section 7.1.3, a time-space span must be either busy or idle, but not both. This concept can be used to measure efficiency.

### 7.3.3 Efficiency

The efficiency of a linear processor pipeline can be defined as the ratio of time-space span required by the taskforce against the total time-space span, which equals the sum of all busy and idle time-space spans. If $n, k$, and $\tau$ are the number of samples, number of pipeline stages and sample period, respectively, the efficiency of a LPP architecture $\mu$ is:

$$
\begin{equation*}
\mu=\frac{n k \tau}{k[k \tau+(n-1) \tau]}=\frac{n}{k+(n-1)} \tag{7-17}
\end{equation*}
$$

Note that $\mu \rightarrow 1$ as $n \rightarrow \infty$, implying that the larger the number of samples flowing through the pipeline, the higher its efficiency, as Figure 7-4. From Equation (7-16) it is apparent that $\mu=S(k) / k$. Consequently, $\mu$ can be interpreted as the ratio of actual speed-up to the ideal unitary speed-up $k$. In steady-state with $n » k$, the efficiency tends to 1 .


Figure 7-6: Efficiency characteristics for LPP M-Ps.

### 7.3.4 Throughput

Throughput is defined as the number of tasks that can be completed per unit time. This rate reflects the compute power of a given M-P architecture. In terms of efficiency $\mu$ and sample period $\tau$ the throughput for a linear processor pipeline can be defined as:

$$
\begin{equation*}
w=\frac{n}{k \tau+(n-1) \tau}=\frac{\mu}{\tau} \tag{7-18}
\end{equation*}
$$

where $n$ is equals the total number of samples being processed during the observation period $k \tau+(n-1) \tau$. In the ideal case, $w=1 / \tau=f_{s}$ when $\mu \rightarrow 1$. Maximum throughput is therefore equal to the sampling frequency, corresponding to one output per sample period.

### 7.4 Task Granularity

M-P system architects have long debated the relative merits of fine and coarse granularity. Accordingly, several M-P performance models are developed with respect to this metric.

### 7.4.1 Introduction

In this section, it is assumed that each task executes in $r$ instruction cycles, and that IPC takes $c$ cycles when tasks are not co-resident. Various studies have shown that M-P performance depends strongly on $r / c$. This ratio expresses how much IPC overhead is incurred per unit of computation and, as such, is a measure of computation task granularity [Stone, 1989].

In coarse-grained parallelism, $r / c$ is high, tasks are relatively large and IPC overhead can be amortised over many instruction cycles. Conversely, in fine-grained taskforces the overhead per unit computation $r / c$ is relatively low. Many more PEs may be efficiently utilised than in coarse-grained schemes, but not without significant uplift in IPC overhead.

### 7.4.2 Non-Overlapped IPC

In DMCs, the central aim is to execute a taskforce consisting of $m$ tasks efficiently on $n$ processors. In this section, M-P architectures with non-overlapped IPC are considered.

### 7.4.2.1 Two-Processor Case

In order to minimise IPC overhead when using a dual-DSP M-P, all $m$ tasks can be assigned to one DSP and the other ignored. Alternatively, tasks can be allocated across both in any combination. Total processing time $\omega(m, n, k)$ must then include activities such as IPC which incur time overhead. When $k$ tasks are assigned to one PE , and $m-k$ to the other, then:

$$
\begin{equation*}
\omega(m, n, k)=r \max (m-k, k)+c(m-k) k \tag{7-19}
\end{equation*}
$$

Note that the second term models IPC transfers. One optimum solution, illustrated in Figure 7-7, is to assign all tasks to one processor if $r / c$ is less than or equal to $m / 2$. If the computation per unit communication exceeds this threshold, then the tasks should be split evenly as Figure $7-8$. That is, either $k_{o p t}=0$ or $m$ for $r / c \leq m / 2$, or $m / 2$ otherwise.


Figure 7-7: Non-overlapped IPC with 50 tasks, where $r / c=10: k_{o p t}=0$ or $m$.

### 7.4.2.2 Extension to $\boldsymbol{n}$ Processors

In a M-P with $n$ processors available for taskforce execution, Equation (7-19) becomes:

$$
\begin{align*}
\omega(m, n, k) & =r \max \left(k_{i}\right)+\frac{c}{2} \sum_{i=1}^{n} k_{i}\left(m-k_{i}\right) \\
& =r \max \left(k_{i}\right)+\frac{c}{2}\left(m^{2}-\sum_{i=1}^{n} k_{i}^{2}\right) \tag{7-20}
\end{align*}
$$

where $k_{i}$ is the number of tasks assigned to the $i$-th processor.
From Section 7.4.2.1, the difference in processing costs of 'even' ${ }^{1}$ distribution and uni-processor execution, $\Delta \omega$, is given by:

$$
\begin{equation*}
\Delta \omega(m, n, k)=\frac{r m}{n}+\frac{c m^{2}}{2}-\frac{c m^{2}}{2 n}-r m \tag{7-21}
\end{equation*}
$$

where the first three terms are the cost of even allocation.
Ignoring values of $m$ that are not exact multiples of $n$, Equation (7-21) solves for $r / c$ :

$$
\begin{equation*}
\frac{r}{c}=\frac{m}{2} \tag{7-22}
\end{equation*}
$$

[^34]

Figure 7-8: Non-overlapped IPC with 50 tasks, where $r / c=40: k_{o p t}=m / 2$.
If $r / c>m / 2$ then even distribution will produce the best schedule. If, however, $r / c$ is below or equals this threshold then as $n \rightarrow \infty$ no assignment produces a faster schedule than $n$.

### 7.4.2.3 Speed-Up

Control of overhead costs is absolutely essential for DCS to be successful. The overhead per unit computation $r / c$ determines the point at which parallelism is cost-effective. Even when $r / c$ is sufficiently high to warrant parallelism, the performance gain is diminished by the second term in Equation (7-20). The speed-up $S(n)$ is then:

$$
\begin{align*}
S(n) & =\frac{r m}{\left(\frac{r m}{n}+\frac{c m^{2}}{2}-\frac{c m^{2}}{2 n}\right)} \\
& =\frac{\frac{r n}{c}}{\left(\frac{r}{c}+\frac{m(n-1)}{2}\right)} \tag{7-23}
\end{align*}
$$

If $m$ and $n$ are small and $r / c$ large, then $S(n) \propto n$. However, if the number of processors is increased such that $n \rightarrow \infty$, then $S(n)=2 r / c m$. Hence, $S(n)$ does not depend on $n$ and approaches a constant asymptote as $n \rightarrow \infty$. Even though performance can be improved incrementally as $n$ increases, diminishing returns defeat this M-P strategy.

### 7.4.3 Fully Overlapped IPC

In practice, IPC can be successfully overlapped with computation to minimise overhead penalties as described in the previous chapter. Such a M-P model is developed here.

### 7.4.3.1 Two-Processor Case

If overhead can be overlapped fully with execution costs then Equation (7-20) becomes:

$$
\begin{align*}
\omega(m, n, k) & =\max \left(r \max \left(k_{i}\right), \frac{c}{2} \sum_{i=1}^{n} k_{i}\left(m-k_{i}\right)\right) \\
& =\max \left(r \max \left(k_{i}\right), \frac{c}{2}\left(m^{2}-\sum_{i=1}^{n} k_{i}^{2}\right)\right) \tag{7-24}
\end{align*}
$$

While this model is perhaps over-optimistic, the design of HYMIPS (see Chapter 6) shows that it is in fact possible to fully overlap computation with IPC and other overhead.


Figure 7-9: Fully overlapped IPC execution of 50 tasks, where $r / c=15: k_{\text {opt }}=3 m / 10$.

The threshold occurs when the execution time is just long enough to completely mask the overhead incurred by IPC, as illustrated in Figure 7-9, i.e. $r(m-k)=c(m-k) k$, or:

$$
\begin{equation*}
k_{o p t}=\frac{r}{c} \tag{7-25}
\end{equation*}
$$

where $k$ is restricted to $1 \leq k \leq m / 2 . \omega(m, n, k)$ becomes $r(m-r / c)$, and the speed-up:

$$
\begin{equation*}
S(n)=\frac{1}{1-\frac{r}{c m}} \tag{7-26}
\end{equation*}
$$

$S(n)$ lies between 1 and $n$ and is again maximised when $r / c=m / 2$. As $r / c \rightarrow 1$, $S(n) \rightarrow 1$ and the optimum distribution becomes more unbalanced. This M-P model also depends on $r / c$, but gives rather more optimistic performance predictions.

### 7.4.3.2 Extension to $\boldsymbol{n}$ Processors

For $n$ processors and for any given maximum value of $k_{i}$, even distribution of tasks produces minimum IPC overhead. Hence, the best possible $\omega$ for fully overlapped IPC occurs when:

$$
\begin{equation*}
\frac{r m}{n}=\frac{c m^{2}}{2}\left(1-\frac{1}{n}\right) \tag{7-27}
\end{equation*}
$$

which simplifies to:

$$
\begin{equation*}
\frac{r}{c}=\frac{m(n-1)}{2} \tag{7-28}
\end{equation*}
$$

For larger $n$, Equation (7-28) shows that $n$ for minimum $\omega$ is given by:

$$
\begin{equation*}
n=\frac{2 r}{c m} \tag{7-29}
\end{equation*}
$$

As $m$ increases, the best strategy is to use increasingly fewer processors. For $n=2$, $r / c=m / 2$ which is consistent with previous findings. That $n$ decreases with $m$ is the result of IPC overhead increasing $m$-times faster than the total execution cost, $\omega$.

### 7.4.4 Linear IPC Overhead

Assuming linear IPC overhead, IPC costs are proportional to $n$ rather than the number of tasks assigned remotely. If each task communicates $\alpha$-times with a task on each of the $n$ PEs:

$$
\begin{equation*}
\omega(m, n, k)=r \max \left(k_{i}\right)+\alpha c n \tag{7-30}
\end{equation*}
$$

As only the first term depends on $k$, $\omega$ is mimimised by distributing the $m$ tasks evenly.
As $n \rightarrow \infty$, the increase in $\alpha c n$ eventually becomes larger than the decrease in $r \max \left(k_{i}\right)$. Correspondingly, there is a maximum value of $n$ for which performance increases:

$$
\begin{equation*}
\frac{d \omega(m, n, k)}{d n}=-\frac{r m}{n^{2}}+\alpha c \tag{7-31}
\end{equation*}
$$

which becomes zero when:

$$
\begin{equation*}
n=\sqrt{\frac{r m}{\alpha c}} \tag{7-32}
\end{equation*}
$$

This square-root ensures that the cost of incrementing $n$ cannot be justified as $n \rightarrow \infty$.

### 7.4.5 Multiple Communication Links

A common assumption in the previous models is that IPC proceeds sequentially. It is perfectly possible, as discussed in Chapters $5 \& 6$, to replicate architectural features for IPC. ${ }^{1}$

### 7.4.5.1 Non-Overlapped IPC

If the number of communication links is allowed to increase with $n$ to provide full interconnection, then communication operations can be overlapped. In so doing, $c$ is no longer a constant but itself becomes a function of the number of processors, $n$. However, even with $O\left(n^{2}\right)$ links installed, no more than $O(n)$ concurrent IPC messages can be supported:

$$
\begin{align*}
\omega(m, n, k) & =r \max \left(k_{i}\right)+\frac{c}{2 n} \sum_{i=1}^{n} k_{i}\left(m-k_{i}\right) \\
& =r \max \left(k_{i}\right)+\frac{c}{2 n}\left(m^{2}-\sum_{i=1}^{n} k_{i}^{2}\right) \tag{7-33}
\end{align*}
$$

In Equation (7-33), it is assumed that a processor is either computing, communicating, or idle, and that the total cost of IPC decreases inversely with $n$. From Section 7.4.2, it is known that the previous equation is minimised by assigning tasks as evenly as possible. Under such a schedule, Equation (7-33) becomes:

$$
\begin{equation*}
\omega(m, n, k)=\frac{r m}{n}+\frac{c m^{2}}{2 n}\left(1-\frac{1}{n}\right) \tag{7-34}
\end{equation*}
$$

[^35]
### 7.4.5.2 Threshold Analysis

Parallel execution is beneficial until $\omega(m, n, k)$ fails to decrease as $n \rightarrow \infty$, i.e. $\delta \omega=0$.

$$
\begin{align*}
\delta \omega(m, n, k) & =\frac{r m+\frac{c m^{2}}{2}}{n(n+1)}-\frac{\frac{c m^{2}}{2}(2 n+1)}{[n(n+1)]^{2}}  \tag{7-35}\\
& =\left[r+\frac{c m}{2}\left(1-\frac{2}{n}\right)\right] \frac{m}{n(n+1)}
\end{align*}
$$

However, Equation (7-35) is positive for $n \geq 2$, and so $\omega(m, n, k)$ improves for nearly all $n$. Comparing U-P processing cost with that for an $n$-PE architecture gives the threshold:

$$
\begin{equation*}
\frac{r}{c}=\frac{m}{2 n} \tag{7-36}
\end{equation*}
$$

In this case, the task granularity factor $r / c$ and the number of processors, $n$, are inversely related: the larger $n$ becomes, the smaller the task granularity required. As a result, this M-P is a failure at break-even. The performance provided by $n$ processors is identical to that of one, yet real cost is increased by a factor of $O(n)$ processors and $O\left(n^{2}\right)$ links.

### 7.4.5.3 Fully Overlapped IPC

If IPC overhead can be fully overlapped with execution and $\omega$ minimised by assigning tasks as evenly as possible as before, then the finishing-time $\omega$ becomes:

$$
\begin{equation*}
\omega(m, n, k)=\max \left(\frac{r m}{n}, \frac{m^{2}}{2 n}\left(1-\frac{1}{n}\right)\right) \tag{7-37}
\end{equation*}
$$

This minimum value of $\omega$ occurs when computation fully overlaps IPC overheads:

$$
\begin{equation*}
\frac{r m}{n}=\frac{c m^{2}}{2 n^{2}}(n-1) \tag{7-38}
\end{equation*}
$$

corresponding to a computation to communication ratio of:

$$
\begin{equation*}
\frac{r}{c}=\frac{m}{2}\left(\frac{n-1}{n}\right) \approx \frac{m}{2} \tag{7-39}
\end{equation*}
$$

As $r / c$ is virtually independent of $n$, it is only contingent on the size of DMC taskforce. Providing that the task granularity exceeds the threshold shown in Equation (7-39), this configuration is a success as $S(n)=n$. If, however, $r / c \leq m / 2$, computation does not overlap IPC, so M-P performance is reduced as speed-up is constrained by $m$ :

$$
\begin{equation*}
S(n)=\frac{2 r}{c m} n \tag{7-40}
\end{equation*}
$$

### 7.5 Conclusions

The model developed here has sufficient structure to exhibit many of the difficulties encountered in DCS. Console taskforces are described by the number of tasks $m$ and their resource requirements, with ordering restrictions represented by precedence relations. The M-P is specified by the number of processors $n$, their resources and the capacity of IPC links.

In addition, a DCS algorithm must observe all precedence relations, ensuring that no task can be started until all its predecessors are finished. Due to the indivisible nature of ASP kernels it must also be non-preemptive: a task cannot be interrupted until completion. Since parameters are known in advance, the DMC can be efficiently scheduled at compile-time.

Returning to the DMC context, the conventional speed-up metric described in Section 7.3.3.1 assumes that the amount of parallel and sequential work in the taskforce are fixed. If, however, this ratio changes (see Figure 7-4) as the underlying M-P and taskforce are scaled together SSU indicates how performance will vary for alternative DMC configurations.

If $w_{p}(n) / w_{s}(n)$ is constant, then conventional speed-up and SSU curves are identical. If, however, the ratio of $n$-fold parallel to sequential work increases with $n$, then $S(n)>S^{\prime}(n)$ for $1 \leq n \leq m$ and $S(n)<S^{\prime}(n)$ for $n>m$. Conversely, if $w_{p}(n) / w_{s}(n)$ decreases as $n$ increases, then $S^{\prime}(n)$ lies above $S(n)$ for $1 \leq n \leq m$ but below it when $n>m$. ${ }^{1}$

Conventional speed-up quantifies the relative performance of a DMC taskforce with a fixed ratio of parallel to serial code on a $n$-PE M-P to that of an identical U-P. In situations were the taskforce and underlying M-P are to be scaled, SSU provides a more accurate insight. Depending on how the ratio changes with $n$, SSU can exceed conventional speed-up.

As noted in Section 3.2.3.3, $k_{\text {max }}=48$ for a LPP-based DMC. Equation (7-16) implies that as $n \rightarrow \infty, S(k)=k_{\text {max }}$, i.e. unitary, provided that no more than $k_{\text {max }}$ PEs are required. In real-time $n » k$, and Equation (7-4) shows that efficiency $\mu \rightarrow 100 \%$, provided that $\tau$ instructions are allocated to each PE. In this scenario, throughput $w$ equals $f_{s}$.

[^36]Analysing the LPP shows that this approach to parallelising a DMC taskforce provides opportunities to achieve unitary speed-up and maximum processing efficiency with a throughput of one sample per sample period. This technique has been demonstrated within the HYMIPS architecture described in Chapter 6 and will appear again in the context of DCS.

In Figure 7-10, speed-up behaviour is plotted for M-Ps with a single communications channel. With non-overlapping IPC, $r / c>m / 2$ ensures that even distribution is more effective than allocating all tasks to a single DSP. As the DMC is scaled speed-up improves incrementally, but reaches a constant asymptote of $2 \mathrm{r} / \mathrm{cm}$ irrespective of other factors.


Figure 7-10: Speed-up characteristics with a single IPC link: $r / c=200, m=50$.

Fully-overlapping IPC with task execution gives unitary speed-up, $S(n)=n$, until IPC overhead overwhelms execution costs. Once the threshold of $n=2 r / \mathrm{cm}+1$ is reached, however, $S(n)$ tends towards the same constant asymptote. A DMC with linear IPC overhead performs well until the linear increase in IPC costs progressively limits this approach.

The effect of multiple IPC links on M-P performance are shown in Figure 7-11. With non-overlapped execution, Equation (7-36) confirms that for this case even distribution for any $n$ is more effective than U-P allocation. Overlapping task execution with IPC is the ideal M-P configuration, giving unitary speed-up providing that task granularity exceeds $m / 2$.


Figure 7-11: Speed-up characteristics with multiple IPC links: $r / c=200, m=400$.
A number of different M-P models have been investigated with respect to task granularity. In every case the role of $r / c$ is the same: small ratios lead to higher IPC overhead. For maximum performance, granularity must be offset against this overhead. Unitary speed-up can be achieved by overlapping computation with multiple IPC links.
$\omega$ is totally dependent on IPC when run-time is smaller than IPC overhead. Consequently, it is essential that IPC overhead is restricted to be no greater than execution time. For each model, there is some maximum $n$ that is cost-effective: this number depends on the M-P topology, on the underlying IPC mechanisms and the particular DMC taskforce.

### 7.6 References

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## Chapter 8

## Scheduling Strategies

Scheduling is one of the few areas of discrete optimisation that is motivated by practical considerations. Research in this broad area has been ongoing since the early 1960s. In this chapter, the author develops this discipline with specific regard to the DMC problem domain.

### 8.1 Combinatorial Optimisation

An overview of combinatorial optimisation (CO) is presented in this section. Emphasis is on the fundamental issues defining the intrinsic complexity of the DMC task scheduling problem.

### 8.1.1 Overview

A wide variety of important CO problems have emerged from such diverse areas as operations research and VLSI design. Solving such problems amounts to finding the optimal solution among an extremely large, but finite, number of alternatives. Optimisation algorithms can be used to find globally optimal solutions to a CO problem, often in a prohibitive amount of time.

Approximation algorithms yield solutions in an acceptable amount of time at the risk of sub-optimality. General algorithms are applicable to a wide variety of problems, whereas tailored algorithms rely on problem-specific information. Performance analysis concentrates on the quality of the final solution and run-time required for average and worst-case scenarios.

### 8.1.2 Definitions

Algorithm time requirements are conveniently expressed in terms of a single variable - the 'size' - which reflects the amount of input data needed to describe a problem. The input length, $l$, is defined as the number of symbols used to describe a problem instance in the chosen encoding scheme. ${ }^{1}$ This measurement is used as a formal indication of instance size.

[^37]Time complexity functions express algorithm time requirements in terms of the largest amount of time needed to solve problem instances of a given size. These functions are not well-defined until the encoding scheme is fixed. However, choices made here have little effect on the broad distinctions made in the theory of computational complexity [Garey, 1978].

### 8.1.3 Encoding Schemes

A taskforce can, for example, be described by listing all tasks and arcs, or by the taskforce adjacency matrix. Referring to Figure 7-1, it is clear that various schemes can give different $l$ for the same console. Fortunately, any algorithm having polynomial-time complexity under one scheme will have polynomial-time complexity under all others [Coffman, 1976].

| Scheme | Encoding | Length | Bound $^{\text {a }}$ |
| :--- | :---: | :---: | :---: |
| task list, <br> arc list | $T_{1} T_{2} T_{3} T_{4} T_{5} T_{6} T_{7}$, <br> $\left(T_{1} T_{5}\right)\left(T_{2} T_{5}\right)\left(T_{3} T_{6}\right)\left(T_{4} T_{6}\right)\left(T_{5} T_{7}\right)\left(T_{6} T_{7}\right)$ | 50 | $2 t+6 a$ |
| neighbour <br> list | $\left(T_{5}\right)\left(T_{5}\right)\left(T_{6}\right)\left(T_{6}\right)\left(T_{1} T_{2} T_{7}\right)\left(T_{3} T_{4} T_{7}\right)\left(T_{5} T_{6}\right)$ | 38 | $2 t+4 a$ |
| adjacency <br> matrix | $\left[\begin{array}{lllllll}0 & 0 & 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 0 & 0 & 1 & 0 \\ 1 & 1 & 0 & 0 & 0 & 0 & 1 \\ 0 & 0 & 1 & 1 & 0 & 0 & 1 \\ 0 & 0 & 0 & 0 & 1 & 1 & 0\end{array}\right]$ |  |  |

Table 8-1: Analysis of alternative encoding schemes for DMC taskforces.
a. where $t$ represents the number of tasks and $a$ the number of arcs in the console taskforce.

In fact, it would be hard to envisage a 'reasonable' encoding scheme that differs more than polynomially from Table 8-1. Although 'reasonable' is not easily formalised, it is clear that the scheme employed in a DCS algorithm should be concise and not padded with unnecessary symbols. Hence, numbers should be represented in any fixed base other than 1 .

### 8.1.4 Time Complexity Functions

Scheduling algorithms posses a wide range of time complexity functions. However, the distinction between polynomial-time and exponential-time algorithms is clear.

### 8.1.4.1 Upper and Lower Bounds

Despite some successes, the record for proving lower bounds on the complexity of specific problems is poor [Garey, 1979]. Time complexity as defined is a worst-case measure or upper bound: a time complexity of $2^{l}$ means only that at least one problem instance of size $l$ requires $O\left(2^{l}\right)$ operations to complete. For example, the FFT, one of the most frequently used signal processing algorithms, requires a maximum of $O(l \log l)$ arithmetic operations.

### 8.1.4.2 Polynomial-Time Algorithms

An important concept is the identification of the class of problems, $P$, solvable in time bounded by a polynomial in the length of the input, $l$. Correspondingly, a polynomial-time algorithm is one whose time complexity function is $O(p(l))$ for some polynomial function $p$. That polynomial-time roughly corresponds to tractability was first expressed by Edmonds who termed polynomial-time algorithms 'good algorithms' [Edmonds, 1965].

### 8.1.4.3 Exponential-Time Algorithms

In enumerative disciplines, as $m \rightarrow \infty$, the number of feasible schedules grows explosively. If a taskforce consists of $m$ tasks then, assuming no precedence, the number of U-P schedules is $m!$, an exponential function in $m$. ${ }^{1}$ Extending this argument to a M-P context, the Prolog source of Figure 8-1 shows that the number of schedules of $m$ tasks on $n$ PEs, $N(m, n)$, is:

$$
\begin{equation*}
N(m, n)=\frac{(m+(n-1))!}{(n-1)!} \tag{8-1}
\end{equation*}
$$

Algorithms with complexity not bounded as Section 8.1.4.2 are termed exponentialtime. Note, this category includes non-polynomial (NP) functions, such as $l^{\log l}$, which are not normally regarded as exponential. Most exponential-time algorithms are merely variations on exhaustive search, whereas polynomial schemes usually demand some deeper insight.

[^38]

```
domains
```



```
    d num_sch = real
    d_tsk_dom = symbol
    d_pro_nam = symbol
    d_tsk_lst = d_tsk_dom*
    d_pro_dom = pr( d_pro_nam, d_tsk_lst)
    d_pro_lst = d_pro_dom*
```



```
database
    sch( d_pro_lst)
```



```
predicates
    runalg( d_tsk_lst, d_pro_lst, d_num_sch)
    schedl( d_tsk_lst, d_pro_lst, d_pro_lst)
    delete( d_tsk_dom, d_tsk_lst, d_tskllst)
    delete( d_pro_dom, d_pro_lst, d_pro_lst)
    insert( d_pro_dom, d_pro_lst, d_pro_lst)
    chkdba( d_pro_lst)
    getsech( d_num_sch, d_num_sch)
```



```
clauses
    runalg( TaskList, ProcList, _):-
        retractall( ),
        schedl( TaskList, ProcList, Schedule),
        assert( sch( Schedule)),
        fail
    runalg( _, ,' NumScheds) : -
        getsch( 0, NumScheds)
    schedl([], Schedule, Schedule):-
        chkdba( Schedule), :.
    schedl( TaskList1, ProcList1, Schedule):
        delete( Task, TaskListl, TaskList2),
        delete( pr( ProcName, Done), ProcListl, ProcList2),
        insert( pr( ProcName, [Task|Done]), ProcList2, ProcList3),
        schedl( TaskList2, ProcList3, Schedule).
    delete( Head, [Head|Tail], Tail).
    delete( Head1, [Head2|Tail1], [Head2|Tail2]):-
        delete( Headl, Taill, Tail2).
    insert( pr( PN1, D1), [Pr( PN2, D2)|T], {pr(PN1, D1), pr(PN2, D2)|T]):-
        PN1 < PN2,:
    insert( pr( ProcName, Done), [Proc|Tail1], [Proc|Tail2]):- !,
        insert( pr( ProcName, Done), Tail1, Tail2).
    insert( Proc, [], [Proc]).
    chkdba(Schedule):-
        retract( sch( Schedule)), !,
        assert( sch( Schedule)),
        fail.
    chkdba( _)
    getsch( NumScheds1, NumScheds):-
        retract( sch( _)), !,
        NumSchede2 = NumScheds1 + 1,
        getach( NumScheds2, NumSchede).
    getsch( NumScheds, NumScheds)
```

Figure 8-1: Prolog source to generate $N(m, n)$, assuming no precedence.

### 8.1.4.4 NP-Complete Algorithms

Many practical CO problems have been proved to be NP-complete. ${ }^{1}$ Such problems are unlikely to be solvable by an amount of computation effort which is bound by a polynomial $p(l)$. It is clear that $P \subset N P$, as illustrated in Figure 8-2. Although still an open problem, ' $P=N P$ ' is every unlikely, since $N P$ contains many notorious combinatorial problems.

1. An extensive classification is given in the well-known book by Garey and Johnson [Garey, 1979].


Figure 8-2: Illustration of the $N P$ class of optimisation problems.
In spite of intensive research effort, no good algorithms are known for problems in $N P$. Therefore, the practical effect of $N P$-completeness is as a lower bound. A proof that a problem is $N P$-complete is a proof that the problem is not in $P$ and therefore intractable. In the absence of a rigorous definition, $P$ may be safely identified with polynomial-bounded algorithms.

### 8.1.5 Complexity of Task Scheduling

Most scheduling problems are $N P$-complete. In fact, even the general 2-processor case presents an $N P$-complete problem. In this section, reported theoretical results are summarised.

### 8.1.5.1 Uni-Processor Bounds

While many U-P problems are NP-complete, many can be solved to optimality [Lenstra, 1978]. For example, scheduling with arbitrary precedence constraints is clearly trivial because every feasible schedule is optimal. Problems with precedence relations forming a tree can be solved in $O(l \log l)$. Furthermore, scheduling unit-length tasks with arbitrary precedence constraints has been shown to be polynomial solvable [Gonzalez, 1978].

### 8.1.5.2 Two-Processor Bounds

The general 2-processor scheduling problem is $N P$-complete [Bruno, 1974]. Scheduling with arbitrary precedence constraints to minimise $\omega$ for unit-length tasks can be solved in $O\left(l^{2}\right)$ [Coffman, 1973]. Garey and Johnson have developed an $O\left(l^{2}\right)$ algorithm for the case where each task has to meet a given deadline [Garey, 1978]. However, the 2-processor problem for 1- or 2-unit tasks with arbitrary precedence constraints is $N P$-complete [Lenstra, 1978].

### 8.1.5.3 Multi-Processor Bounds

Scheduling tasks with precedence relations and individual deadlines on an n-processor M-P, $n \geq 3$, is solvable in $O(\log l)$ [Conway, 1967]. Assuming unit processing times, the $n$-processor problem for tasks with in-tree precedence can be solved in $O(l)$ due to the oftencited algorithm by $\mathrm{Hu}[\mathrm{Hu}, 1961]$. With arbitrary precedence, however, this problem and therefore general DCS become $N P$-complete in the strong sense [Lenstra, 1978].

### 8.2 Stochastic Techniques

Stochastic search techniques are modelled on processes found in nature. These algorithms are often touted as near-optimal general-purpose strategies for notoriously difficult CO problems.

### 8.2.1 Simulated Annealing

Simulated Annealing (SA) is the mathematical analogue of cooling: just as in the natural annealing process, SA associates a cooling schedule with the CO problem to be optimised.

### 8.2.1.1 Overview

First proposed by Kirkpatrick, SA is a stochastic computational technique derived from statistical mechanics [Kirkpatrick, 1983]. The name originates from the strong analogy with the physical annealing of solids: the final 'frozen' configuration being optimal. SA has evolved as a general and robust technique for solving many $N P$-hard optimisation problems.

```
M = number of moves to attempt;
T = current "temperature";
for (m = 1 to M )
{
    generate a random move i.e. perturb task schedule;
    evaluate the change in energy (cost), \DeltaE;
    if ( }\Delta\textrm{E}<0
    {
        // downhill move towards minima: accept it
        accept move, and update schedule;
    }
    else
    {
        // uphill move: accept maybe
        accept with probability p = e e}|E/T\mathrm{ ;
        update schedule if accepted;
    }
}
```

Figure 8-3: Pseudo-code listing of the Metropolis Algorithm.

To avoid entrapment at local minima, SA employs a stochastic relaxation technique. The Metropolis Algorithm of Figure 8-3 generates a sequence of $M$ configurations for each value of $T$ [Metropolis, 1953]. Without this counter-intuitive feature of accepting 'up-hill' transitions, illustrated in Figure 8-4, SA would revert to a form of iterative improvement.


Figure 8-4: Sketch demonstrating the need for uphill moves.

### 8.2.1.2 Asymptotic Behaviour

It can be shown that, asymptotically, SA behaves as a true optimisation algorithm [von Laarhoven, 1987]. However, optimal configurations are only guaranteed for an infinite number of transitions [Lundy, 1986]. Convergence, and hence efficiency, depend on the socalled 'cooling schedule'. Aarts and Korst have derived a polynomial-time cooling schedule which can approximate asymptotic behaviour for some problem instances [Aarts, 1989].

### 8.2.1.3 Performance

SA finds near-optimal solutions which do not depend strongly on initial configuration [Aarts, 1989]. Computational effort for a given quality of solution is predominantly dependent on cooling schedule, which in term must be determined by trial and error. Furthermore, average-case run-time is typically close to worst-case [Gaudiot, 1988]. Experience shows that for even basic CO problems, SA is outperformed by tailored heuristics [Johnson, 1987].

### 8.2.2 Genetic Algorithms

Another popular stochastic search technique, genetic algorithms (GAs) are based on a model of population genetics and the principles of the Darwinian Theory of Natural Selection.

### 8.2.2.1 Overview

Holland has shown that a population of bit-strings can be made to "evolve as populations of animals do" [Holland, 1975]. If effective genetic operators can syntactically manipulate such chromosomal representations, then GA can be exploited as Figure 8-5. Populations constitute a parallel search trajectory through the space of all possible solutions. Chromosomes that survive from generation to generation are those which prove to be most fit.

```
P = population of chromosomal representations;
T = number of generations to attempt;
t = current configuration;
initialise population P(0);
evaluate population P(0);
for ( t = 1; t < T &\alpha !solution; t++ )
{
    // get most fit population
    select P(t) from P(t-1);
    // recombine population
    crossover P(t);
    mutate P(t);
    invert P(t);
    // determine fitness
    evaluate P(t);
}
```

Figure 8-5: Outline of GA procedure.

### 8.2.2.2 Recombination Operators

Recombination operators introduce new genetic material into the population by guiding the generation of new chromosomes through the modification of bit-string structures.

## - Crossover

The function of crossover is to combine beneficial genetic material from parent chromosomes in the next generation. For example, the operator in Table 8-2 takes two structures, splits the strings at the same randomly determined point, and then creates two new structures by swapping the tail portions. It is this feature of evolution which is the key to the power of GAs.

| Old Structures |  | New Structures |  |
| :---: | :---: | :---: | :---: |
| 100 | 10110 | 100 | 00101 |
| 111 | 00101 | 111 | 10110 |

Table 8-2: Example behaviour of GA crossover operator.

## - Mutation

Mutation randomly changes a bit in a structure so introducing new material into the knowledge base. Assigned a very low percentage of activation, this operator occasionally introduces beneficial material into a problem instance. As a result, mutation operators guarantee that no genetic configuration will be permanently lost from a population.

## - Inversion

Altering the positional linkages of bits within one structure by inverting the sequence between two randomly assigned points is termed inversion. For instance, ' 110011101 ' randomly cut after the 2 nd and 6th bits, gives '111100101' when the centre sub-string is inverted. In this way, the location of genes is changed to provide a more fertile ground for recombination.

### 8.2.2.3 Performance

Like natural genetic systems, GAs quickly locate optimal regions in vast search spaces [Holland, 1975]. Unfortunately, many implementations are prone to premature convergence. Careful implementation of operators is necessary to adjust the balance between exploration and exploitation [Booker, 1987]. Many application-dependent issues, such as the percentage of descendants created by each operator, have considerable effect on optimality.

### 8.2.3 Application to DCS

While both paradigms have been successfully implemented for a range of difficult CO problems, their application to DCS is mitigated by algorithm- and problem-dependent issues.

### 8.2.3.1 Simulated Annealing

Although, asymptotically, SA is capable of locating near-optimal solutions, average-case algorithm run-time is typically close to worst-case. Furthermore, both solution and run-time
are largely dependent on cooling schedule. Due to these facts, a DCS based on SA would require a prolonged empirical study in an effort to determine suitable cooling schedules.

While polynomial-time cooling schedules may approximate asymptotic behaviour, a DMC control system would have to rely on pre-determined cooling schedules. No bounds can be determined on the quality of schedules generated in the real production environment. In addition, work in other areas has shown that SA is outperformed by tailored heuristics.

### 8.2.3.2 Genetic Algorithms

Unfortunately, GA implementations are often prone to premature convergence due to the over-exploitation of individual operators. To increase the ratio of exploration and exploitation requires a judicious balance of the descendants created by each operator. Again algorithmic issues have a direct bearing both on the run-time of the algorithm and quality of the solution.

Opportunities exist for incorporating problem-specific information into GAs: firstly, the initial schedule may be determined by a bootstrap heuristic; secondly, recombination can go beyond simple syntactic processing. However, DCS does not map well onto this numerical scheme and non-probabilistic techniques are better suited to exploiting tailored heuristics.

### 8.3 Deterministic Strategies

The static scheduling problem is not unique to the DCS problem being considered here. As it also exists in many other guises a variety of deterministic solutions have been proposed.

### 8.3.1 Graph Theory

As early as the 1960s, Ford and Fulkerson proposed a graph theoretic approach to task scheduling [Ford, 1962]. Subsequently, Stone showed that the problem could be solved by using algorithms constructed for finding maximum flows in commodity networks [Stone, 1973]. Developing this work, Lo included the concept of interference costs [Lo, 1988]. Although helpful for $n \leq 3$, this technique has limited application to DCS.

### 8.3.2 Integer Programming

Task scheduling can be cast as integer programming by defining a suitable objective function. To this end, Chu suggested a optimisation structure of the following type [Chu, 1980].

### 8.3.2.1 Problem Construction

$$
\begin{equation*}
\omega_{o p t}=\sum_{i=1}^{m} \sum_{j=1}^{n}\left[w x_{i, j} s_{i, j}+(1-w) \sum_{k=1}^{m} \sum_{1=1}^{n} d_{j, l} s_{i, j} s_{k, l}\right] \tag{8-2}
\end{equation*}
$$

where $w(0 \leq w \leq 1)$ is the relative weighting of the two terms. The first term of Equation (8-2) is simply the execution time of $T_{i}$ on $P_{j}$. The second term sums the IPC delays across all tasks that must communicate with $T_{i}: d_{j, l}$ is the IPC delay from $P_{j}$ to $P_{l}$. Finally, Equations (8-3) and (8-4) model memory and execution time constraints for each PE:

$$
\begin{align*}
& \sum_{i=1}^{m} m_{i} s_{i, k} \leq M_{k}, \text { where } k=1, \ldots, n \\
& m  \tag{8-3}\\
& \sum_{i=1}^{m} x_{i, j} s_{i, k} \leq X_{k}, \text { where } k=1, \ldots, n
\end{align*}
$$

### 8.3.2.2 Application to DCS

This 01 integer-programming approach has several liabilities, notably considerable computational expense [Hillier, 1986]. In terms of DCS, the absence of precedence relations is a critical shortcoming. Another drawback is the difficulty involved in extending this technique to include the finite capacity of IPC channels. Although it is possible to include additional constraints, run-time problems are greatly exacerbated [Thesen, 1978].

### 8.3.3 Clustering Techniques

Rather than tailor a problem to a mathematical technique, it may be more natural to begin with problem characteristics and develop a mapping algorithm, as shown by Efe [Efe, 1982]. Following this scheme requires developing an algorithm that not only clusters tasks and balances workload, but also selects a number of nodes and assigns clusters to these nodes.

Such clustering and iterative reassignment may arrive at solutions quickly for certain problems. However, the method is not always able to manage all constraints and can become
trapped in local minima [Konstantinides, 1989]. Although preferable to integer programming, this technique shares many of the former's problems and has limited application to DCS.

### 8.3.4 Dynamic Programming

Dynamic programming is a mathematical technique employed in problems where finding a solution can be reduced to a sequence of interrelated decisions forming a path in a state-space.

### 8.3.4.1 Overview

Scheduling taskforces on M-P architectures can be naturally cast in terms of state-space searching. As shown in the illustration of Figure 8-6, each alternative decision from a node corresponds to scheduling a different task, or idle period, on a processor. At each step the nonintuitive option of scheduling an idle period, lasting until the next processor has completed its currently assigned task, must be allowed to guarantee optimality [Coffman, 1976].


Figure 8-6: Task scheduling in terms of state-space searching.

### 8.3.4.2 Basic Strategies

Two basic strategies for exploring state-spaces are depth-first and breadth-first search: depthfirst leads to long run-times as the search is uninformed; breadth-first disciplines maintain the set of decision nodes on the growing edge of the search tree. With an objective function as the optimisation criteria, best-first methods expand the best partial path at each node.

Expanding the best partial path at each decision node reduces algorithm run-times dramatically, but does not reduce the combinatorial explosion exhibited in state-space searching. A more sophisticated scheme is required, using problem-specific information. Heuristic rules avoid aimless paths, drawing the search process quickly towards a goal node.

### 8.3.4.3 $\quad A^{*}$ Search

As A* is effectively an improved branch-and-bound, it is the preferred paradigm for the task scheduling problem examined here. $A^{*}$ employs the best-first principle with a heuristic underestimate of remaining cost [Nilsson, 1982]. It is based on the intuitive notion that a optimal solution is more likely to be reached by following paths with lower underestimates.

For each decision node $n$ corresponding to a partial schedule, $h^{*}(n)$ denotes the cost of an optimal path to a goal node. A heuristic $h(n)$ is admissible if for all $n$ if [Pearl, 1985]:

$$
\begin{equation*}
h(n) \leq h^{*}(n) \tag{8-5}
\end{equation*}
$$

A* employs $h(n)$ to guide the search tree expansion. One of the algorithms principle features is that it never expands a node with an underestimate greater than $\omega_{\text {opt }}$ [Sinclair, 1987].

### 8.3.4.4 Application to DCS

Although $A^{*}$ can revert to exhaustive search worst-case, alternative heuristics can be investigated by observing their ability to prune the search-tree. In order to realise tailored deterministic algorithms, four admissible application-specific heuristics are defined below.

## - Minimum Processor Cost - MPC

This heuristic corresponds to the optimistic estimate of the finishing time of the partial schedule, completed with all currently unassigned tasks. If $t=\left\{t_{1}, t_{2}, \ldots, t_{m}\right\}$ is the list of un-scheduled tasks and $p=\left\{p_{1}, p_{2}, \ldots, p_{n}\right\}$ the list of active processors, then:

$$
\begin{equation*}
\mathrm{MPC}=\frac{1}{n}\left(\sum_{i=1}^{m} e\left(t_{i}\right)+\sum_{j=1}^{n} f\left(p_{j}\right)\right)-F(t, p) \tag{8-6}
\end{equation*}
$$

where $e\left(t_{i}\right)$ is the execution time of task $t_{i}, f\left(p_{j}\right)$ the partial finishing time of processor $p_{j}$, and $F(t, p)$ the finishing time of the partial schedule. Note that MPC is computed under the assumption that precedence, preemption and distributed execution constraints are relaxed.

## - Maximum Remaining Critical-Path - MRC

Alternatively, the finishing time of the remaining critical path for the most recently assigned task on each processor can be evaluated with respect to the current partial schedule. If $s l\left(t_{i}\right)$ is the level of task $t_{i}, f t\left(t_{i}\right)$ the finishing time of $t_{i}$, and $F(t, p)$ the finishing time of the partial schedule as before, then MRC for any node in the $A^{*}$ search-tree can be defined as:

$$
\begin{equation*}
\operatorname{MRC}=\max \left(s l\left(t_{i}\right)-e\left(t_{i}\right)+f t\left(t_{i}\right)-F(t, p)\right), \quad 1 \leq i \leq m \tag{8-7}
\end{equation*}
$$

## - Minimum Independent Assignment - MIA

The minimum assignment cost for each of the remaining tasks is computed ignoring all unscheduled tasks, but including IPC with all non-coresident (NCR) tasks already assigned. Here, $i\left(t_{i}\right)$ represents the earliest time that all NCR IPC is available to $t_{i}$ on the earliest ready processor. MIA is then defined as the sum of minimum assignment costs for all $t$, or:

$$
\begin{equation*}
\mathrm{MIA}=\sum_{i=1}^{m}\left(i\left(t_{i}\right)+e\left(t_{i}\right)-F(t, p)\right) \tag{8-8}
\end{equation*}
$$

## - Maximum Heuristic Underestimate - MHU

By definition, MPC, MRC and MIA lie within the constraints imposed by Equation (8-5). If all underestimates are calculated simultaneously for each node, then MHU can be defined as:

$$
\begin{equation*}
\mathrm{MHU}=\max (\mathrm{MPC}, \mathrm{MRC}, \mathrm{MIA}) \tag{8-9}
\end{equation*}
$$

As the scheduler search-tree expands, clearly the individual heuristics approximate $h^{*}(n)$ with varying degrees of accuracy, particularly when $n$ is in the vicinity of a goal node.

### 8.4 List Scheduling

In list scheduling strategies, tasks are first ordered according to some external priority scheme and then scheduled according to non-increasing priority.

### 8.4.1 Critical Path Method

One technique used to form a priority list is the critical path method (CPM), originally investigated by $\mathrm{Hu}[\mathrm{Hu}, 1961]$. In what is probably the most frequently cited reference in M-P scheduling, Hu developed two problems for equal-duration tasks. Although only optimal when precedence constraints form a rooted tree, the concept of Hu -level is widely applicable.

In CPM, as in PERT analysis, tasks are assigned to PEs according to the length of the precedence chains they head. The longest chains in the unexecuted part of the taskforce are those which have the greatest sum of execution times: they are termed 'critical paths' because the tasks which lie on them are those most likely to retard the execution of the DMC taskforce.

### 8.4.2 Hu-Level Labelling

A task $T_{i}$ is given the label $s l\left(T_{i}\right)=l p\left(T_{i}\right)+1$, where $l p\left(T_{i}\right)$ is the length of the longest path from $T_{i}$ to the root task, which is labelled ' 0 '. Tasks one unit removed are labelled ' 1 ', as Figure 8-7. As DMCs contain many root tasks', a 'dummy' root facilitates labelling. Priority is assigned by non-increasing Hu -level and the taskforce scheduled according to this regime. ${ }^{2}$

Ties are broken by assigning $T_{i}$ to $P_{j}$ with the smallest index $j$ or, alternatively, by scheduling $T_{i}$ with the longest execution time. Tasks with $e\left(T_{i}\right)>1$ can be represented by an indivisible chain of unit-weight tasks whose sum equals $e\left(T_{i}\right)$. It can be shown that the level of the chain-task furthest from the root is equal to that of the multi-unit task [Kohler, 1975].


Figure 8-7: Hu-level labelling of a taskforce consisting of unit-length tasks.

[^39]
### 8.4.3 Alternative Labelling Schemes

The basis of the list scheduling discipline suggests several different labelling techniques. Assigning priorities assuming all tasks have identical $e\left(T_{i}\right)$, corresponds to Highest Levels first with No estimated times (HLN). Under this taxonomy, conventional CPM as described in Section 8.4.1 corresponds to Highest Levels first with Estimated times (HLE).

If a task's co-level is measured from the entry rather than exit task, two more labelling schemes are implied. Lowest Co-levels first with Estimated times (LCE) gives a task with the smallest co-level highest priority. If all tasks are assumed to have the same execution time and the LCE concept applied, this gives Lowest Co-levels first with No estimated times (LCN).

### 8.4.4 Upper Bounds on Schedule Quality

Hu's optimal solutions are limited to rooted trees of tasks with equal execution time. In this section, other known theoretical upper bounds for list schedules are compared and contrasted.

### 8.4.4. 1 Preemptive vs. Non-Preemptive

List schedules may be preemptive or non-preemptive. For arbitrary $n(n \geq 1)$, if the length of a non-preemptive schedule is $\omega_{n}$ and the length of a preemptive (optimal) schedule $\omega_{p}$ then: ${ }^{1}$

$$
\begin{equation*}
\frac{\omega_{n}}{\omega_{p}} \leq 2-\frac{1}{n} \tag{8-10}
\end{equation*}
$$

The upper bound of Equation (8-10) can be shown to be tight for $n=2$ [Coffman, 1973].
Kohler has shown that pathological problems for which $\omega_{n} / \omega_{p}$ is arbitrarily close to $3 / 2$ can be constructed for any $n \geq 2$ [Kohler, 1975]. He also conjectured that:

$$
\begin{equation*}
\frac{\omega_{n}}{\omega_{p}} \leq 2-\frac{2}{n+1} \tag{8-11}
\end{equation*}
$$

if CPM is used to produce schedules, though this tighter bound has not yet been proven.

### 8.4.4.2 Precedence Constraints

When there are precedence constraints among the tasks, the worst-case performance bound for CPM with unit-length tasks, that is, the ratio of $\omega_{c p m}$ to the optimal finishing-time, $\omega_{o p t}$, is:

[^40]\[

$$
\begin{equation*}
\frac{\omega_{c p m}}{\omega_{o p t}} \leq 2-\frac{1}{n} . \tag{8-12}
\end{equation*}
$$

\]

When there are no precedence constraints, the tasks are distributed across the processors purely by the order given in the priority-list $L$ [Graham, 1976].

It has been shown that, for a set of independent tasks, specifically:

$$
\begin{equation*}
\frac{\omega_{c p m}}{\omega_{o p t}} \leq \frac{4}{3}-\frac{1}{3 n} \tag{8-13}
\end{equation*}
$$

which is substantially less than Equation (8-12), as $n \rightarrow \infty$. Since there are instances where these bounds are attained, they cannot be improved. As Equation (8-12) crosses the $y$-axis at the origin, it is better than Kaufman's bound for small values of $\omega_{\text {opt }}$. [Kaufman, 1974].

### 8.4.4.3 Unequal Task Times

Let $\omega_{p}$ be the length of a taskforce scheduled by an algorithm which ignores the task uninterruptibility condition. Let $\omega_{\text {opt }}$ be the length of an optimal schedule, under the restriction that tasks may not be interrupted. If $\omega_{c p m}$ is the length of schedule given by the CPM algorithm (which also may not interrupt tasks) then, after Coffman [Coffman, 1973]:

$$
\begin{gather*}
\omega_{p} \leq \omega_{o p t}  \tag{8-14}\\
\omega_{o p t} \leq \omega_{c p m} \tag{8-15}
\end{gather*}
$$

Equation (8-14) follows from the fact that the possible sequence of schedules by any restricted algorithm are a subset of those available to the optimal unrestricted algorithm. Equation (8-15) follows from the definition of an optimal algorithm, since CPM is also restricted its schedules are possible outcomes for the algorithm which found $\omega_{\text {opr }}$.

### 8.4.4.4 Mutually Commensurate Task Times

The results discussed above can be extended to trees providing that task times are mutually commensurate [Kaufman, 1974]. Tasks with weights greater than one can be represented by an indivisible string of unit-weight tasks. As all tasks in the restructured taskforce are of unit length, the optimal non-preemptive solution can be found using Hu's algorithm.

If $\omega_{p}$ is the length of an optimal schedule with arbitrary preemption, and $\omega_{c p m}$ that determined by a CPM algorithm prohibiting preemption of indivisible string tasks, then:

$$
\begin{equation*}
\omega_{c p m} \leq \omega_{P}+k-\left\lceil\frac{k}{n}\right\rceil \tag{8-16}
\end{equation*}
$$

where $k$ is an upper bound on task size [Kaufman, 1974]. With $\omega_{o p t}$ representing the optimal length of any non-preemptive schedule, combining Equation (8-14) to Equation (8-16) gives:

$$
\begin{equation*}
\omega_{p} \leq \omega_{o p t} \leq \omega_{c p m} \leq \omega_{p}+k-\left\lceil\frac{k}{n}\right\rceil \tag{8-17}
\end{equation*}
$$

### 8.4.5 Application to DCS

In many instances critical-path schedules are extremely efficient. Figure 8-8 illustrates that even for worst case the theoretical finishing time, $\omega_{c p m}$ lies within twice the optimal value $\omega_{\text {opt }}$ for unit task times. Clearly a good strategy for constructing feasible DMC schedules, this heuristic scheduling strategy can be implemented to run in $O(n \log m)$ [Garey, 1978].


Figure 8-8: Upper bounds for CPM with unit task times.

Taskforces with arbitrary precedence and mutually commensurate $e\left(T_{i}\right)$ can be scheduled efficiently, providing that $k$ is restricted as Figure 8-9. Incidentally, this discipline is more likely to generate an optimal solution as $n \rightarrow \infty$, for fixed $m$. As $\omega$ approaches the lower limit determined by the critical-path, increasing $n$ will not reduce $\omega$ further.


Figure 8-9: Upper bounds for CPM with mutually commensurate task times.

### 8.5 M-P Scheduling Anomalies

It has been shown, both theoretically and in practice, that scheduling algorithms can often locate feasible schedules which are at odds with intuitive concepts.

### 8.5.1 Idle Periods

With preemptive disciplines, it is never beneficial to introduce idle periods on processors when there are tasks available for execution. This is not true with non-preemptive disciplines as illustrated by the taskforce of Figure 8-8 and schedules of Figure 8-9. When $P_{2}$ is idle in the optimal schedule $S_{1}$ from 2 to $4, T_{7}$ is continuously available for assignment. By disallowing idle periods and assigning $T_{7}$ to $P_{2}$ at time 2, the longer schedule $S_{2}$ is obtained.

### 8.5.2 List Scheduling

In list scheduling strategies, tasks are usually scheduled in order according to some external priority scheme. Schedule length depends not only on $n$, but also on the priority list used.

### 8.5.2.1 Overview

The unpredictable or anomalous behaviour of list scheduling refers to those instances when counterintuitive increases in schedule length $\omega$ are produced by: increasing the number of


Figure 8-10: Example taskforce to illustrate the idle time anomaly. ${ }^{\text {a }}$
a. Here, the symbol ' $x$ ' represents the execution time of the adjacent task.
processors in $P$; decreasing some of the execution times in $T$; relaxing some of the precedence constraints in <; or by using a different priority list, $L$. That is, efforts intended to improve the values of model parameters can cause an unavoidable increase in $\omega$.


Figure 8-11: Schedules of the taskforce of Figure 8-8 to illustrate the idle time anomaly.

### 8.5.2.2 Upper Bounds

Graham has developed a general limit on these adverse effects by executing a set of tasks twice [Graham, 1976]. During the first execution tasks are characterised by the set of parameters $\{T,<, L, P, \omega\}$ and during the second execution by $\left\{T,<^{\prime}, L^{\prime}, P^{\prime}, \omega^{\prime}\right\}$ such that every constraint of $<$ is also in $<$. The result of this general bound is:

$$
\begin{equation*}
\frac{\omega^{\prime}}{\omega} \leq 1+\frac{n-1}{n^{\prime}} \tag{8-18}
\end{equation*}
$$

Graham has shown that this bound is the best possible and, for instances where the number of processors remains the same (i.e. $n=n^{\prime}$ ), the inequality has the elegant form:

$$
\begin{equation*}
\frac{\omega^{\prime}}{\omega} \leq 2-\frac{1}{n} \tag{8-19}
\end{equation*}
$$

which is achieved by the variation of any one of $L, T$, or $<$. As examples exist which match this inequality, the upper bound on the right-hand side cannot be improved [Graham, 1979].

### 8.5.2.3 Alternative Priority Lists

When 3 processors are used the inequality of Equation (8-20) is never more than $5 / 3$. Even utilising the worst case priority list increases the finishing-time compared with that obtained using the best possible list by no more than $66.67 \%$. This performance guarantee holds irrespective of the complexity of a given taskforce. No matter how carefully or carelessly the priority list is chosen, the ratio of finishing-times is strictly bounded by Equation (8-20).

### 8.5.2.4 Reduced Execution Times

Anomalies that result when task execution times are reduced were first discussed by Richards [Richards, 1960]. Results of simulations show that approximately $80 \%$ of test cases exhibit such irregularities. Manacher has developed an algorithm such that the completion time of a taskforce is not increased by reducing the execution time of any task. This is accomplished by adding a modest number of precedence constraints to the original ordering [Manacher, 1967].

### 8.6 Conclusions

In this analysis, consideration has been given to some of the more prominent aspects of scheduling deterministic DMC taskforces. It is assumed that taskforce precedence graphs are
acyclic, with no branching and that task execution times are exactly known a priori. It should be noted, though, that in other DSP environments these assumptions are not necessarily valid.

With the complexity of the DCS problem derived in Equation (8-1), the borderline between polynomial and $N P$-complete problems is apparent. Among these are found important examples such as scheduling unit-length tasks with arbitrary precedence for $n \geq 3$, and taskforces with in-tree precedence constraints, which have particular relevance here.

Both SA and GA techniques require empirical tuning of implementation-specific factors to guarantee convergence to an optimal solution. Depending on a cooling schedule or recombination operators to generate feasible console schedules within a run-time acceptable in the studio environment makes these stochastic disciplines somewhat unpromising for DCS.

Referring back to Figure 7-1, it is clear that the problem domain under investigation contains rather more inherent structure - e.g. precedence - than those in which general optimisation performs well. $A^{*}$ dynamic programming enables tailored heuristics, such as MPC, MRC, MIA and MHU developed in Section 8.3.4.4 to leverage directly off this structure.

Whereas $A^{*}$ can revert to exhaustive search, list scheduling forms the basis of many non-enumerative disciplines. This is particularly true as guaranteed upper bounds can be derived for many algorithms which fall into this category. Within the bounds discussed here, the choice of method used to generate the priority-list can make a great difference to $\omega$.

Intuition suggests that the best priority lists are those in which tasks on longest critical paths appear near the head. In that case only relatively small additions to processor usage would be made at the end of the schedule. Four such labelling schemes - HLN, HLE, LCN and LCE - are put forward in Section 8.4.3 and will be developed further in the next chapter.

Lack of idle time and non-preemption are responsible for the unpredictable behaviour of CPM scheduling. Ready processors are forced to begin the execution of relatively unimportant tasks: tasks that are short or involved in few precedence constraints. Execution cannot be interrupted to commence more urgent tasks that subsequently become available.

By determining the performance ratio for an approximation algorithm, information is obtained which provides a useful and rigorously defined quantity with which different
approximation algorithms can be compared. In the remainder of this thesis, the author shows how tailored heuristic disciplines capture the salient features of scheduling DMC taskforces.

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## Chapter 9

## DCS Research Framework

Details are given of the framework developed by the author to support the investigation of taskforce scheduling disciplines. Major features of the $D C S$ engine are examined and results reported to illustrate the application of $\mathrm{A}^{*}$ and CPM disciplines to digital mixing products.

### 9.1 Software Environment

In order to research scheduling algorithms in the DMC problem domain, a software environment has been developed specifically to harness the $D C S$ deliverable. Incidentally, the methodology used here retains much of the philosophy developed at SSL for the 01 console.


Figure 9-1: Block diagram of $D C S$ research framework.

### 9.1.1 Hierarchical Console Design

A schematic is created from symbols forming its internal, functional description. The part also has a symbolic description, which represents its external appearance. The symbol is a box, or other appropriate shape, with input and output links as required by the schematic. Symbolic descriptions may be placed, or 'instanced', into any schematic higher up the DMC hierarchy.

Hierarchical design techniques lead to a clear, well-structured constructs. They facilitate 'top-down' design, starting with the overall functional requirement and adding detail as the design develops. A properly structured design ensures that common ASP tasks are debugged in isolation and entered only once, thus simplifying product development.

### 9.1.2 Graphics Capture

As illustrated in Figure 9-1, a hierarchical graphics-capture package has been employed to enable console taskforces to be entered as schematic diagrams. For example, an EQ section may be treated as a block in a symbol - such as a channel strip, and may itself be a network of ASP tasks - such as low, high, and mid-frequency sections (see Section 2.2.3).


Figure 9-2: Symbol representing a 2-input bus-configured mix task.

In the same manner, the description of the target hardware may be generated via graphics-capture. To this end, a library of schematic primitives has been assembled as illustrated in Figures 9-2 \& 9-3. Any DMC taskforce or M-P architecture that can be realised with primitives provided by the console manufacturer may be prepared for scheduling.


Figure 9-3: Symbol representing HYMIPS Motorola DSP56001 module.

### 9.1.3 Behavioural Models

Behavioural models have been created in the HELIX simulation language for each of the primitives in the DMC library. As shown in Figure 9-4, symbol attributes indicating resource requirements can be passed as parameters to these models. Of course, this approach could be extended to ASP in order to replace the trace routines currently used for debugging purposes.

As shown in Figure 9-5, a bi-quad kernel requires 14 instruction cycles, 7 words of P-space for code storage, and a further 10 of D-space for filter coefficients. A 27 MHz DSP56001 provides 281 instruction cycles per 48 kHz sample period, up to 192 K words of $\mathbf{P}$ and D RAM, and can support 4 analogue or direct digital input-output ports [Motorola, 1989].

### 9.1.4 Taskforce Simulation

A data-flow simulator can be generated from behavioural models of the console ASP tasks, in accordance with the structure present in the taskforce schematic [Linton, 1991]. Two types of simulation can then be performed: symbolic, to verify the connectivity of the ASP algorithms in the DMC taskforce; and timed, to obtain an ideal lower bound on the schedule length.

Such taskforce simulation was employed in the early stages of this research to verify the operation of the scheduling algorithms investigated in this thesis. A similar approach has been used to test the three IPC schemes supported by the predicates sch_non_cor (. . . ) and sch_ins_ipc (Ipc, ...), described in detail later in this chapter in Section 9.2.3.1.


```
MODULE TkfBhv;
```



```
USE External Modules;
NETTYPE audiobus = RECORD
    audio : bite24;
    trace : PtrToString
    END
(* --------------------.--------------------------------------------------*)
TASKTYPE Mix( instruction cycles : integer; (* attributes passed from... *)
            memory_words : integer); (* ...graphics capture symbol *)
    INWARD
            ip1, ip2 : audiobus
    OUTWARD
            OP : audiobus,
    VAR
            TaskName : string20;
            TaskNum : integer;
            working : audiobus;
BEGIN
    IF InitialisationPhase THEN BEGIN (* Initialisation Code *)
            NEW( op.trace);
            op.audio := 0
            Op.trace^ := "";
        END
        ELSE BEGIN (* Simulation Code *)
            IdentifyTask( TaskName, TaskNumber);
            EnableInputActivationTest( ip1);
            EnableInputActivationTest ( ip2);
            REPEAT
            (* main body of behavioural model *)
            WAITFOR InputActivation( ip1) AND InputActivation( ip2)
                                    CHECK ip1, ip2;
            working.audio := new sample value;
            UpdateTrace( TaskName, ipl.trace, ip2.trace, working.trace);
            ASSIGN working TO op DELAY instruction_cycles;
        UNTIL FALSE; (* UNTIL simulation ends *)
    END; (* ELSE *)
END; (* Mix *)
```



```
behavioural models for other taskforce primitives
```



Figure 9-4: Module of HELIX behavioural models for DMC taskforce primitives.

### 9.1.5 Preliminary Results

As an example of inital investigations, the DMC taskforce shown in Figure 9-6 can be executed sequentially such that $\omega_{s c h}=136$. With a critical path length $\omega_{c p}$ of 62 cycles, this 4:1 console can be allocated optimally onto 4-DSPs as Figure 9-7. Mean processor efficiency is $54.85 \%$ although $P_{1}$ remains unallocated throughout. While both $\mathrm{A}^{*}$ and CPM disciplines produce an optimal schedule, $A^{*}$ requires 4-times the run-time of the latter method.


Figure 9-5: DSP56001 assembly code for DMC EQ biquad kernel.

### 9.2 Digital Console Scheduler

As shown in Figure 9-1, a digital console scheduler is required in order to allocate a DMC taskforce of $m$ tasks onto a M-P hardware architecture comprising $n$ processors. Here, the predicate hierarchy of the $D C S$ engine is described in relation to supported menu options. ${ }^{1}$

### 9.2.1 Overview

In this overview, the implementation of $D C S$ is described together with the specification of the computer system used in this thesis. Database encoding schemes are also presented.

[^41]

Figure 9-6: Example 4-input: 1-output console taskforce.


Figure 9-7: Optimal allocation of example taskforce of Figure 9-6.

### 9.2.1.1 Implementation

The scheduling algorithms outlined in the following pages have been coded in Borland's Turbo Prolog, with executables running on a 66 MHz Pentium with 16 Mb RAM. As a
compiled Prolog, this implementation proves efficient though it should be noted that the code generator targets the 80286. No advantage is taken of the '586 enhanced instruction set.

As revealed in Section 7.1, the DMC taskforce consists of ASP tasks with execution times known a priori and precedence constraints - represented here as nets in a schematic. The Turbo Prolog Toolset, which provides facilities similar to UNIX lex and yacc utilities, has been used to parse the BNF netlists produced by the SilvarLisco tools into clauses.

### 9.2.1.2 Database Encoding

As examined in Section 8.1.3, any 'reasonable' encoding scheme ensures that polynomialtime complexity can be maintained. Within the internal Prolog database, the taskforce is represented using the data model detailed in Table 9-1. After the database is consulted, these structures are resolved into the internal $D C S$ representation described in dcs_glb.pub.

| Description | Domain $^{a}$ |
| :--- | :--- |
| task | task (d_tsk_nam, d_tsk_typ) |
| task type | tasktype(d_tsk_typ, d_tsk_exe, d_tsk_inp, d_tsk_out) |
| net | net(d_net_nam, d_net_inp, d_net_out) |

Table 9-1: $D C S$ internal database representation of DMC taskforces.
a. The prefix $d_{-}$indicates that the corresponding symbol is a user-defined domain name.

For efficiency reasons, non-numeric domains are represented as symbols, where possible. Note that $d_{-} n e t$ inp is a Prolog symbol whereas $d_{-} n e t$ _out is a d_sym_lst, as free-grouping enables a DMC channel to operate as a broadcast signal path. Similarly, the homogeneous M-P architectures employed in this thesis are denoted as shown in Table 9-2.

| Description | Domain |
| :--- | :--- |
| processor | proc (d_pro_nam, d_pro_typ) |
| processor type | proctype (d_pro_typ, d_pro_exe) |

Table 9-2: $D C S$ internal database representation of M-P architectures.

### 9.2.2 Scheduling Strategies

The three distinct algorithms developed as part of this research project are reviewed in the following sections. For example, selecting CPM from the scheduling strategy menu causes $D C S$ to employ a critical-path list-scheduling discipline to schedule the DMC taskforce.

### 9.2.2.1 Critical-Path Method

Choosing CPM initiates a list scheduling algorithm developed from the work reported in Section 8.4.1 and detailed as pseudo-code in Figure 9-8. cpm_bgn_alg (Ipc, Lbl, ...) calls utl_lbl_tkf (Lbl, . . .) to label the taskforce using the Lbl strategy. ${ }^{1}$ Using a quick-sort method, utl_srt_prl (. . .) produces a priority list ordered by non-increasing level. As Table D-1 indicates, this implementation supports five static labelling schemes.


```
// DCS CPM scheduling algorithm
```



```
cpm_bgn_alg(...)
{
    // label taskforce
    initialise levels to zero;
    current_task = root_task;
    while ( !entire taskforce traversed)
    {
        // calculate HLE label
        select next_task;
        new_level = current_task.level + next_task.execution_time;
        // update task level
        if ( new_level > next_task.level )
            next_task.level = new_level;
    }
    // create priority list
    quick-sort task-list by non-increasing level;
    // bchedule taskforce
    while ( priority list != [] )
    {
            // select task with no unfinished predecessors
            while ( task has unfinished predecessors )
                select next task from priority list;
            // schedule ready task on available processor
            if ( available processor has sufficient resources )
                schedule task on this processor;
    }
}
//
```



Figure 9-8: Pseudo-code of $D C S$ CPM algorithm, using the HLE labelling scheme.

### 9.2.2.2 Dynamic List Scheduling

dyn_bgn_alg(Ipc, Lbl, Dyn, ...) extends CPM by dynamically re-labelling the taskforce after each scheduling step. In order to store the dynamic levels of ready-task,

[^42]available-processor combinations, dyn_ini_edb() initialises a 4-th order B-tree in EMS. dyn_cre_sch (Ipc, Dyn, ...) interrogates the B-tree and schedules tasks accordingly. As dynamic labelling schemes are the subject of Chapter 10 , they are not detailed further here.

### 9.2.2.3 A* Search

To initialise the search-tree, ast_ini_tre ( . . .) creates a EMS-resident B-tree with a 'null' root node. ast_exp_tre (Ipc, Hrt, ...) generates all successors $n$ of the cheapest live node bound by ast_chp_liv (...), computes $h(n)$ and inserts $n$ in order of non-decreasing cost. If ast_gol_nod (...) locates a goal node which satisfies Equation (8-5), AST completes. Otherwise ut __rep_eat () generates an infinite supply of backtracking points.

### 9.2.3 Control Parameters

During operation DMC taskforce and M-P architecture are selected via BGI list boxes. $D C S$ control parameters are chosen from the modeless menu structure presented in Table D-1.

### 9.2.3.1 IPC Mechanism

The work in Section 7.4 indicates that several IPC mechanisms are relevant here. inf_get_ipc (...) accepts user input and inf_vld_ipc (. . .) validates the menu item against known IPC schemes. The option chosen is passed as xxx_bgn_alg(Ipc, ...).

- FWE - fully overlapped with execution

Calling sch_ins_ipc (Ipc, ...) with Ipc bound to s_ipc_fwe schedules multiple IPC messages fully-overlapped with task execution as described in detail in Section 7.4.5.3. ${ }^{1}$

- NSL — non-overlapped, single link

Applying sch_ins_ipc (Ipc, ...) with Ipc bound to s_ipc_nsl assumes one IPC link and a transport mechanism that cannot overlap with task execution (see Section 7.4.2.2).

- NML - non-overlapped, multiple links
sch_ins_ipc (s_ipc_nml, ...) facilitates multiple IPC channels but, as advanced in Section 7.4.5.1, concurrent computation and communication is not supported in this instance.

[^43]
### 9.2.3.2 Static Labelling Scheme

$D C S$ supports the four static labelling schemes described in Section 8.4.3, in addition to a random strategy. inf_get_lbl (. . .) accepts user input and inf_vld_lbl (. . .) validates the menu item. The labelling scheme is passed to CPM as cpm_bgn_alg (Ipc, Lbl, ...).

## - HLN - highest levels first, no estimated times

Calling utl_lbl_tkf (s_lbl_hln, ...) with Lbl bound to s_lbl_hln assigns levels using utl_lbl_all (Lbl, ...) which assumes all tasks have identical execution time.

## - HLE - highest levels first, estimated times

Conventional CPM as described in Section 8.4 .1 corresponds to $u t l_{\_} 1 b l_{\_} t k f(L b l, \ldots)$ with Lbl bound to s_lbl_hle which is subsequently passed to utl_lbl_all (Lbl, ...).

## - LCN - lowest co-levels first, no estimated times

Defining task co-level as measured from the entry rather than exit task, utl_lbl_tkf (s_lbl_lon, ...) gives a task with small co-level bigh priority.

## - LCE - lowest co-levels first, estimated times

utl_lbl_tkf(s_lbl_lce, ...) applies the lowest co-level concept using utl_lbl_all (s_lbl_lce, ...) which estimates task times using utl_est_exe (....).

- RND - random
utl_lbl_tkf( $\mathrm{Lbl}, \ldots$ ) with Lbl bound to s_lbl_rnd assigns arbitrary task labels, calling utl_ini_lvl(s_ini_rnd) to initialise levels randomly in the range 0 to 100 .


### 9.2.3.3 Dynamic Labelling Scheme

dyn_bgn_alg (Ipc, Lbl, Dyn, ...) initiates the DYN scheduling algorithm with both static and dynamic taskforce labelling schemes. inf_get_dyn (. . . ) accepts user input and inf_vld_dyn (. . . ) validates the choice against known $D C S$ dynamic-labelling schemes.

- NON - none

NON is essentially a conventional statically-labelled CPM scheme, employing dyn_ins_dis (s_dyn_non, ...) to insert static levels into the DYN priority mechanism.

- PRO - select available processor first
dyn_ins_dls(Dyn, ...) with Dyn bound to s_dyn_pro inserts dynamic levels for all ready tasks $T_{r}$ using the available processor $P_{a}$ with smallest partial finishing-time.


## - TSK - select ready task first

dyn_ins_dls(s_dyn_tsk, ...) generates dynamic levels for the ready task $T_{r}$ which has the highest static level for all available processors $P_{a}$ in the partial schedule.

- ALL - all combinations

With Dyn bound to s_dyn_all, dyn_ins_dls (Dyn, . . .) inserts dynamic levels for all $T_{r}$ and $P_{a}$. dyn_sch_nxt ( $\ldots$ ) schedules $T_{r}$ on $P_{a}$ such that dynamic level is maximised .

### 9.2.3.4 Heuristic Underestimate

Four heuristics admissible in the DMC problem domain have been described in Section 8.3.4. inf_get_hrt (. . ) accepts user input and inf_vld_hrt (. . .) validates the menu item chosen. The heuristic underestimate is passed to AST as ast_bgn_alg (Ipc, Hrt, ...).

- NON - none

Calling ast_hrt_val(Hrt, ...) with Hrt bound to s_hrt_non simply returns zero, corresponding to no heuristic underestimate or $O\left(n^{m}\right)$ enumeration of all feasible solutions.

- MPC - minimum processor cost
ast_hrt_val (s_hrt_mpc, Ipc, ...) estimates the finishing time of the partial schedule via ast_mpc_hrt (Ipc, ... ) which relaxes all the constraints of Section 7.1.3.


## - MIA - minimum independent assignment

ast_hrt_val(s_hrt_mia, Ipc, ...) calls ast_mia_hrt(Ipc, ...) which ignores all unscheduled tasks but includes IPC with all non-coresident (NCR) tasks already assigned.

- MRC - maximum remaining critical-path
$\omega$ for the remaining critical-path for the most recently assigned task on each PE is evaluated by ast_hrt_val(s_hrt_mrc, Ipc, ...) through calling ast_mrc_hrt(Ipc, ...).


## - MHU - maximum heuristic underestimate

As MPC, MRC and MIA must all satisfy Equation (8-5), ast_hrt_val (s_hrt_mhu, ...) gives MHU by calculating all heuristics simultaneously and binding to the maximum value.

### 9.2.4 Scheduler Outputs

As well as displaying a textual description of the DMC schedule, including idle periods and IPC transfers, $D C S$ also supports console taskforce and M-P architecture analysis.

### 9.2.4.1 Pre-Schedule Statistics

Using lst_num_elm (. . . ) on the task-list, precedence-list and processor-list created from internal database facts provides the number of tasks $m$, precedence relations and processors $n$, respectively. In order to quantify the distribution of task granularity, clc_tsk_grn (. . .) determines the minimum, maximum and average granularity of tasks in the DMC taskforce.

Sequential schedule length $\omega_{\text {seq }}$ is calculated in clc_seq_sch(...) by simply summing all task execution times. clc_crt_pth (. . ) determines the critical path $\omega_{c p}$ by temporarily labelling the taskforce using HLE. Taskforce parallelism, defined as $\omega_{s e q} / \omega_{c p}$, acts as lower bound on $n$ required to minimise the schedule length, $\omega_{s c h}$.

### 9.2.4.2 Pre-Schedule Checks

In the $D C S$ implementation presented in this thesis, three separate indicators are used to establish that the DMC taskforce can be scheduled on the selected M-P architecture. Firstly, inf_grn_tst (. . .) recursively determines if all the ASP tasks in the DMC taskforce have execution times strictly less than the processing resource supplied by each of the M-P PEs.
inf_fit_tst (. . . ) binds to s_tst_pas if the hardware resource capacity given by clc_hdw_rsc (...) is greater than the sequential schedule $w_{\text {seq }}$ or s_tst_fai otherwise. Finally, inf_ctp_tst (...) employs clc_crt_pth (. . .) to ascertain if the critical path inherent in the DMC taskforce is less than the processing resource supplied by each PE.

### 9.2.4.3 Post-Schedule Statistics

Post-schedule statistics are sent to the default output device using the wri_pst_stt (. . .) predicate. clc_run_tim (. . .) calculates algorithm run-time by retracting internal database facts which $\log$ system time prior to $D C S$ execution and on termination. Schedule length $\omega_{\text {sch }}$ is obtained from the completed schedule - the processor-list —by clc_fin_tim(...).

[^44]To calculate speed-up, clc_tsk_spd:(...) uses clc_seq_sch(...) described in Section 9.2.4.1 to determine $\omega_{\text {seq }}$ with $\omega_{\text {sch }}$ obtained as above. clc_thr_put (...) evaluates through-put as the number of ASP tasks computed per sample period. Following the work on efficiency, clc_pro_utl (. . .) determines processor utilisation as Section 7.3.3.

### 9.3 Performance of AST

Here, $D C S$ is used to compare the four admissible underestimates defined in Section 8.3.4.4. A homogeneous 2-DSP56001 target and fully-overlapped (FWE) IPC help determine which of these application-specific heuristics may form the basis of efficient approximation algorithms.

### 9.3.1 Time \& Space Complexity

As discussed previously in Section 8.1.4, the DCS problem can be solved through exhaustive enumeration of all feasible schedules. Equation (8-1) indicates that this scheme is $O\left(n^{m}\right)$ and thus prohibitive for relatively small instances. In order to gauge time complexity, the principal performance metric employed here is the run-time $t$ required to produce an optimal schedule.

Comparisons on this basis alone, however, are biased against MHU since this heuristic requires rather more computation than the individual underestimates also considered in this survey. As a result, the number of state-space nodes $s$ generated by each heuristic is recorded since the ability to prune the search tree gives a strong indication of true space complexity.

### 9.3.2 Algorithm Run-Time

Table 9-3 details AST algorithm run-time for taskforces based on traditional mix-buses. While MRC results in the shortest run-time, relative performance is more visible on $\log -\log$ axes.

### 9.3.2.1 Bus-Configured Taskforces

From Figure 9-9, it is apparent that MPC performs little better than no heuristic underestimate (i.e. NON) for bus-oriented taskforces. Referring back to Equation (8-5), this behaviour is explained by the fact that, in calculating MPC, the remaining unscheduled tasks are simply apportioned across the M-P. As all the constraints of Section 7.1.3 are ignored, $h(n) « h^{*}(n)$.

| channels | NON | MPC | MIA | MRC | MHU |
| :---: | :---: | :---: | :---: | :---: | :---: |
| 4 | $0: 00: 00.06$ | $0: 00: 00.06$ | $0: 00: 00.06$ | $0: 00: 00.06$ | $0: 00: 00.06$ |
| 8 | $0: 02: 27.91$ | $0: 01: 34.27$ | $0: 00: 00.11$ | $0: 00: 00.09$ | $0: 00: 00.08$ |
| 12 | $1: 54: 23.00$ | $0: 48: 14.10$ | $0: 00: 00.93$ | $0: 00: 00.16$ | $0: 00: 00.22$ |
| 16 | $17: 59: 50.86$ | $6: 04: 21.00$ | $0: 00: 11.04$ | $0: 00: 00.28$ | $0: 00: 00.50$ |
| 24 | - | - | $0: 02: 15.01$ | $0: 00: 01.16$ | $0: 00: 01.70$ |
| 32 | - | - | $0: 09: 54.52$ | $0: 00: 03.19$ | $0: 00: 04.50$ |
| 40 | - | - | $0: 22: 36.66$ | $0: 00: 07.69$ | $0: 00: 10.38$ |
| 48 | - | - | $0: 54: 17.63$ | $0: 00: 15.10$ | $0: 00: 20.16$ |
| 56 | - | - | $1: 28: 47.38$ | $0: 00: 26.47$ | $0: 00: 34.88$ |
| 64 | - | - | $2: 24: 00.00$ | $0: 00: 46.19$ | $0: 00: 59.98$ |

Table 9-3: $D C S$ run-time using AST-FWE to schedule bus-type taskforces.

$\log (c)$
Figure 9-9: AST-FWE run-time for bus-type taskforces on a 2-DSP M-P.
While MIA initially appears attractive, this scheme degenerates towards an enumerative process as the number of channels $c$ is increased. Rather better results are reported for MRC and the combined MHU strategy. MHU lies approximately $3.35 \%$ above MRC due to the calculation of MPC, MIA and MRC heuristics for each node in the search-space.

### 9.3.2.2 Tree-Configured Taskforces

Very few results can actually be produced for taskforces with mix-tree structures as shown in Figure 9-10. In fact, AST without any heuristic underestimate ${ }^{1}$ does not complete with 24:00:00.00 for an 8-channel console. While MRC is reasonably efficient in this context, MIA generates optimal schedules in less time for 2 of the 3 cases reported here. With the loose precedence constraints inherent in trees, a combined approach seems to be most effective.


Figure 9-10: AST-FWE run-time for tree-type taskforces on a 2-DSP M-P.

### 9.3.3 State-Space Complexity

While algorithm run-time is one method of comparison, this approach favours those heuristics which are simple to enumerate. Here, they are compared in terms of state-space complexity.

### 9.3.3.1 Bus-Configured Taskforces

As the results displayed in Figure 9-11 show, the behaviour of the AST heuristics in $D C S$ are markedly different when considered in terms of state-space complexity. Although MIA reveals similar unfavourable behaviour as before, MRC and MHU generate exactly the same number of nodes. Both limit state-space expansion equally well when scheduling bus-type taskforces.

Evidently the computation of ast_hrt_val (...) to evaluate Equation (8-9) is directly responsible for the difference in run-times noted in Section 9.3.2.2. NON and MPC

[^45]

Figure 9-11: AST-FWE state-space for bus-type taskforces on a 2-DSP M-P. generate in the order of $10^{5}$ nodes to schedule 16-channel consoles on the 2-PE architecture used here, though the rate of increase is notably less than that apparent in Figure 9-9.

### 9.3.3.2 Tree-Configured Taskforces

Limited results can be produced using AST to schedule taskforces based on tree configurations yet trends similar to those reported in Section 9.3.2.2 are observed. However, instead of MIA lying $9.09 \%$ above MHU for $\log c=1.08$ as shown in Figure 9-12, both heuristics generate the same number of nodes. While not completely conclusive, these results suggest that a judicious combination of MIA and MRC may lead to an effective non-enumerative procedure.

### 9.4 Performance of CPM

Results are reported and compared for the labelling schemes developed in Section 8.4.3, for instances where all IPC is fully overlapped with task execution - FWE. The objective here is to ascertain which of these static schemes prove most suited to the DMC problem domain.

### 9.4.1 Algorithm Run-Time

Since console taskforces may vary between linear constructs and tree-based structures with a high degree of inherent parallelism, $D C S$ run-time is reported for both ends of this spectrum.


Figure 9-12: AST-FWE state-space for tree-type taskforces on a 2-DSP M-P.

### 9.4.1.1 Bus-Configured Taskforces

As illustrated in Figure 9-13, random labelling (RND) proves as effective as any of the 'true' labelling schemes. This rather anomalous behaviour is explained by the very nature of the CPM algorithm. Since the conventional mix-bus is heavily constrained by precedence, only one task is ready at each scheduling step. No advantage is gained by ordering the priority list in advance using utl_srt_prl(...): this process simply adds to computation overhead.


Figure 9-13: CPM-FWE run-time for bus-type taskforces on an 8-DSP M-P.

HLN and LCN are slightly disparate: due to the conversion of task co-levels in ut l_cnv_lvl ( . . ). HLE and LCE extend run-time further as retrieving $e\left(T_{i}\right)$ requires $O(m)$ database queries. Figure $9-14$ re-considers these results from a $\log -\log$ perspective. Ignoring implementation issues, $\log t$ is linear with product-moment correlation $>0.99925$. If $c$ represents the number of channels, linear regression gives: $\log t_{\mathrm{HLE}}=3.866 \log c-10.07$.


Figure 9-14: CPM-FWE $\log (t)$ for bus-type taskforces on an 8-DSP M-P.

### 9.4.1.2 Tree-Configured Taskforces

Repeating this analysis using taskforces configured with mix-trees results in rather different behaviour, however. From Figure $9-15$, it is evident that HLE is extremely effective particularly when compared with HLN and RND schemes. Replotting these experimental results on log-log axes yields piece-wise linear characteristics for the HLE strategy alone.

With LCE and LCN requiring run-times on average $24.15 \%$ higher than HLE, the ordering of decreasing performance - HLE, LCN, LCE, HLN, RND - is evident. Clearly, the additional $O(m)$ overhead involved in estimating task execution in ut1_est_exe (. . .) returns manifest benefits when task priority is not totally constrained by precedence.


Figure 9-15: CPM-FWE run-time for tree-type taskforces on an 8-DSP M-P.

### 9.4.2 Schedule Length

Calculating the ratio of schedule $\omega_{s c h}$ to critical path length $\omega_{c p}$ for a range of (normalised) taskforce parallelism $\pi$ gives the results presented in Figure 9-16. While $\pi<1$, the ideal schedule length $\omega_{c p}$ is achieved, but as $\pi \rightarrow 1, \omega_{s c h}$ increases linearly with $\pi$. This linear relationship is maintained until $\pi>1.55$, when the different schemes start to diverge.


Figure 9-16: CPM-FWE $\omega_{s c h}$ for tree-type taskforces on an 8-DSP M-P.

HLE consistently returns the shortest $\omega_{s c h}$ as $\pi$ increases, surpassing RND by at least $6.86 \%$ for $\pi>1.55$. Interestingly the behaviour of LCN is rather erratic, matching HLE for some values of $\pi$ and yet identical to LCE for $\pi=2.29$. Not surprisingly, RND is the worst strategy for minimising $\omega_{s c h}$ when DMC taskforces contain partial precedence constraints.

### 9.4.3 Speed-Up

From the speed-up characteristics presented in Figure 9-17, it is plain that HLE returns the best performance under this metric. While HLE is close to linear for small $n$, it is ultimately constrained by $\omega_{c p}$ as $n$ is scaled for the 64-channel taskforce employed here. LCN yields identical performance for $n \leq 12$ and $n \geq 40$, though drops to $93.78 \%$ of HLE when $n=40$.


Figure 9-17: CPM-FWE speed-up for tree-type taskforces.

HLN lies below HLE throughout and falls to the level of LCE and RND for $n=24$. RND is the least suitable labelling scheme for maximising speed-up using CPM, except for $n=32$ and $n=40$ when LCE lies $5.26 \%$ and $2.50 \%$ below RND respectively. Although all five strategies converge when $n=56$, LCE is as much as $12.20 \%$ beneath HLE for $n=32$.

### 9.4.4 Scaled Speed-Up

As shown in Figure 9-18, the scaled speed-up measurements fall within one standard deviation for all $n$. While there is little to separate HLN, LCN and LCE, it is evident that HLE is
again the best technique considered in this investigation. RND is typically the worst strategy, confirming the advantage of tailored labelling techniques. In fact, the characteristic obtained with HLE is up to $21.05 \%$ higher than RND for problem instances where $n<48$.


Figure 9-18: CPM-FWE scaled speed-up for tree-type taskforces.

### 9.5 Conclusions

Aspects of the $D C S$ research framework are appropriate to both console design and studio operation. Block-diagram schematics naturally promote the reuse of ASP code and processing hardware, and intuitively describe console layout. Instead of requiring partitioning prior to allocation, LGDF schematics are implicitly partitioned at the level of the ASP task.

The SilvarLisco suite has proved convenient for entering data-flow DMC schematics and simulating their behaviour. A parser, translating HELIX netlists into Prolog clauses, has been written using the Turbo Prolog Toolbox. ${ }^{1}$ Similar netlist manipulation is employed in the design phase to convert M-P architecture schematics assembled from hardware symbols.

In the production environment, users need not be aware of architectural details or even the number of DSPs. While not a central aspect of this thesis, behaviour models developed in

[^46]a regular simulation language have enabled the author to test AST, CPM and DYN algorithms, as well as validate the declarative implementation of NSL, NML and FWE IPC mechanisms.

Internally, the $D C S$ engine retains the current state of the partial schedule together with a list of unscheduled tasks. The scheduler predicate hierarchy is described in Section 9.2 from the user's perspective, with reference to the Prolog source included as Appendix D. If netlist output is supplied, $D C S$ can be interfaced with any schematic-capture package.

Four admissible application-specific heuristics are investigated by comparing the runtime required to obtain an optimal allocation and by observing their ability to prune the $\mathrm{A}^{*}$ search-space. Experience shows that the MRC heuristic is most effective at the start of the scheduling process, with MPC \& MIA proving more effective as the schedule nears completion.

While MPC may be effective towards the end of the process, this underestimate readily causes AST to revert to exhaustive search. Rather better results are reported for MRC and the combined MHU strategy. Although both generate comparable search-spaces, MHU requires significant run-time uplift as three separate $h(n)$ must be determined for each new node $n$.

One aim of this study was to determine if the additional complexity of MIA is justified by a significant reduction in the size of the search-space $s$. With tree-configured taskforces, MIA has a definite run-time advantage over MRC as $\omega_{c p}$ « $\omega_{\text {seq }}$, and $h(n) « h^{*}(n)$ for MRC as a result. Both heuristics limit state-space expansion equally well as $\omega_{c p} \rightarrow \omega_{\text {seq }}$.

MIA reduces the size of the state-space search tree to such a degree when compared to MPC that the additional overhead involved in generating nodes is not significant. Whereas, MIA it is more complex to calculate than MRC, the values of MRC can be effectively precomputed without prior knowledge of the current state of the partial schedule.

Work in Section 8.1.4 has determined that an efficient DCS algorithm is one which requires an amount of time bounded in the input length $l$ by some polynomial. The essentially $O\left(n^{m}\right)$ requirements of enumerative procedures moves the focus of this research towards heuristic approaches, especially methods derived from MRC \& MIA underestimates.

Using the DCS research framework with FWE IPC mechanisms, random-labelling (RND) and the critical path length $\omega_{c p}$ help determine which, if any, static labelling scheme is
well-suited to the DMC problem. Rather than make this comparison with randomly generated taskforces, the author again extracts precedence DAGs from real DMC layouts.

With bus-configured taskforces, no labelling scheme is more effective than simply assigning labels randomly (RND). Since $\omega_{c p}=\omega_{s e q}$, only one ready task exists at any stage in the scheduling process: no improvement is possible by ordering the priority list in advance. Estimating task execution time in HLE and LCE only serves to magnify this effect further.

HLE returns manifest benefits when task priority is not fully constrained by precedence as is the case is a console LGDF displaying in-tree precedence relations. When the sequential schedule $\omega_{\text {seq }}$ is much greater than the length of the critical path, $\omega_{c p}$, the following order of decreasing performance is clearly evident from Figure 9-15-HCE, LCN, LCE, HLN, RND.

As optimal schedules cannot easily be determined, the ratio of schedule to critical path length, $\omega_{s c h} / \omega_{c p}$, allows labelling schemes to be compared in terms of $\omega . \pi$ has a direct bearing on $\omega_{s c h}$ as $n<\pi$ signifies that inherent parallelism will not be fully exploited. With HLE consistently returning the shortest schedule length - HCE, LCN, LCE, HLN, RND.

In terms of speed-up, HLE surpasses other strategies although LCN is as good for small $n$ : HLE, LCN, HLN, RND, LCE prior to convergence. With up to $17.39 \%$ separating HLE and RND schemes in Section 9.4.4, the ordering - HLE, LCN, LCE, HLN, RND - emerges. SSU curves intersect those for speed-up at $n=1$ and $n=m$, as predicted in Section 7.2.4. ${ }^{1}$

CPM-HLE is well inside the theoretical upper bound of Equation (8-17). One flaw in this preferred approach is anomalous behaviour which is a direct result of the static nature of the labelling process. Ready tasks are scheduled onto the earliest available processor, with no regard to the partial schedule. Rectifying this handicap is the main focus of the next chapter.

### 9.6 References

[Graham, 1969] Graham R L: "Bounds on Certain Multiprocessing Anomalies", SIAM Journal of Applied Mathematics, Vol. 17, No. 2, pp. 416-429, March 1969.
[Motorola, 1989] Motorola: DSP56000/DSP56001 Digital Signal Processor User's Manual, Revision 2.0, Motorola Inc., 1989.

[^47]
## Chapter 10

## Dynamic Labelling Schemes

The tendency for AST to revert to an $O\left(n^{m}\right)$ enumerative procedure motivates further research into heuristic approaches. Unfortunately, the static nature of the CPM labelling process is directly responsible for anomalous behaviour. Together, these observations suggest that predetermined task levels should be revised throughout the DMC scheduling process.

### 10.1 Motivation

Before investigating dynamic schemes within the structure of Section 9.4, the author specifies three alternative strategies and defines an upper bound on schedule quality. Demonstrating 'processor thrashing' illustrates some of the advantages engendered by dynamic techniques.

### 10.1.1 Alternative Labelling Schemes

Here, the concept of dynamic-level labelling is defined and the four alternative strategies for task and processor selection, previously introduced in Section 9.2.3, are described in detail.

### 10.1.1.1 Definitions

From Section 8.4.2, $\operatorname{sl}\left(T_{i}\right)$ is the HLE static-level of task $T_{i}$ with $f t\left(P_{j}\right)$ symbolizing the finishing-time of $P_{j}$ in the current partial schedule. If $i a\left(T_{i}, P_{j}\right)$ represents the earliest time at which all IPC required by $T_{i}$ is available on $P_{j}$, then dynamic level may be defined as:

$$
\begin{equation*}
d l\left(T_{i}, P_{j}\right)=s l\left(T_{i}\right)-\max \left(f t\left(P_{i}\right), i a\left(T_{i}, P_{j}\right)\right) \tag{10-1}
\end{equation*}
$$

This expression combines aspects of both MIA and MRC AST heuristics in order to assign highest priority to that ( $T_{i}, P_{j}$ ) incurring minimum extension of the current partial schedule.

### 10.1.1.2 None - NON

NON, effectively a HLE statically-labelled CPM scheme, is included so that implementationspecific issues do not detract from the conclusions presented in this chapter. dyn_clc_lvl (s_dyn_non, ...) simply retrieves $s l\left(T_{i}\right)$ from the internal database and deterministic generator predicates insert static levels into the DYN priority mechanism.

### 10.1.1.3 Select Available Processor First - PRO

In this scheme, the available processor $P_{a}$ with the earliest finishing-time $f t\left(P_{a}\right)$ is first selected by lst_del_elm(Proc, ...) and lst_del_ndt (Task, ...) generates dynamic levels for all ready tasks $T_{r}$. dyn_ins_dls (s_dyn_pro, ...) inserts each $d l\left(T_{r}, P_{a}\right)$ with dyn_sch_nxt (...) scheduling $\left(T_{r}, P_{a}\right)$ with maximal $d l()$.

### 10.1.1.4 Select Ready Task First - TSK

Alternatively, lst_elm_del (Task, . . .) selects the ready task $T_{r}$ with the highest staticlevel $\operatorname{sl}\left(T_{r}\right)$ with the non-deterministic lst_del_ndt (Task, ...) generating dynamic levels for all available processors $P_{a}$. Again, dyn_ins_dls (s_dyn_tsk, ...) inserts each $d l\left(T_{r}, P_{a}\right)$ and dyn_sch_nxt (Dyn, $\ldots$ ) schedules $\left(T_{r}, P_{a}\right)$ as Figure 10-1 below.

### 10.1.1.5 All Combinations - ALL

Finally, dyn_gen_lvl (s_dyn_all, ...) calls the non-deterministic generator predicates lst_del_ndt (Task, ...) and lst_del_ndt (Proc, ...) to construct all possible $\left(T_{r}, P_{a}\right)$. With dyn_ins_dls(s_dyn_all, ...) inserting each $d l\left(T_{r}, P_{a}\right)$ as before, dyn_sch_nxt (Dyn, . . ) schedules $T_{r}$ on $P_{a}$ such that the dynamic level is maximised.

### 10.1.2 Upper Bounds on Schedule Quality

In the preceding discussions in Section 8.4.4 it was assumed that the priority list remains static during the scheduling process. Dynamic labelling schemes seek to redefine the list every time a PE becomes free. If $\omega_{d y n}$ is the finishing time for a taskforce scheduled in this manner and $\omega_{o p t}$ the optimal finishing time then, using the general bound cited earlier:

$$
\begin{equation*}
\frac{\omega_{d y n}}{\omega_{o p t}} \leq 2-\frac{1}{n} \tag{10-2}
\end{equation*}
$$

Graham has developed a slightly better best-possible bound given by [Graham, 1976]:


```
// DCS DYN`scheduling algorithm
```



```
dyn_bgn_alg(...)
{
    // HLE static labelling
    initialise levels to zero;
    this_task = root_task;
    while ( lentire taskforce traversed )
    {
        // calculate HLE level
        new_level = this_task.level + next_task.execution_time;
        // update task level
        if (new_level > next_task.level )
            next_task.level = new_level;
    }
    // create priority list
    quick-sort task-list by non-increasing level;
    // schedule taskforce
    while ( priority list != [] )
    {
        // TSK dynamic labelling
        delete all dynamic levels
        for (this_task = first_task; task <= last_task; i++)
        {
        if (:this_task has unfinished prodecessors)
            {
                for ( processor = first_processor; processor <= last_processor; j++ )
                {
                        if (this_processor has sufficient resources )
                                    // calculate TSK level
                                    dyn_level = this_task.level - max( this_processor.finish_time,
                                    all IPC data available ;;
                }
                    }
        }
        }
        // schedule ready task on available processor
        schedule task-processor with maximum dyn_level;
    }
}
```



Figure 10-1: Pseudo-code of $D C S$ DYN algorithm, using HLE \& TSK labelling schemes.

$$
\begin{equation*}
\frac{\omega_{d y n}}{\omega_{o p t}} \leq 2-\frac{2}{n-1} \tag{10-3}
\end{equation*}
$$

An alternative to this approach is to assign the task whose execution time plus the execution time of all its successors is maximal. If a set of tasks executed in this manner has a finishing time of $\omega_{m}$, then $\omega_{m} / \omega_{o p t}$ is also bounded by the inequality established in Equation (10-2).

### 10.1.3 Processor Thrashing

Scheduling the EQ section from channel 3 in Figure 9-6 on a 2-DSP M-P illustrates the advantages of dynamic labelling in isolation, particularly with regard to processor thrashing.

### 10.1.3.1 Static Taskforce Labelling

With DYN-FWE-HLE-NON, $T_{1}$ is the first task to be scheduled since $s l\left(T_{1}\right)=54$. When $P_{1}$ is replaced in the processor list, it must be placed after $P_{2}$ as $f t\left(P_{2}\right)=0 . T_{3}$ is selected as the next ready task and scheduled onto $P_{2}$. By definition, DYN-FWE-HLE-NON will continue thrashing in this manner until the optimal schedule illustrated in Figure 10-2 is produced,


Figure 10-2: DYN-FWE-HLE-NON schedule for 4-stage EQ section on a 2-DSP M-P.

Scheduling the same taskforce using DYN-NSL-HLE-NON produces the schedule of Figure 10-3. ${ }^{1}$ The explanation of the thrashing exhibited here is similar to that given above, except that IPC must be allocated before $T_{3}, T_{5}$ and $T_{7}$ can be scheduled. Indeed, thrashing is directly responsible for the sub-optimality of this essentially CPM-NSL-HLE strategy.


Figure 10-3: DYN-NSL-HLE-NON schedule for 4-stage EQ section on a 2-DSP M-P.

### 10.1.3.2 Dynamic Taskforce Labelling

Selecting $P_{a}$ first results in the same schedules as above, whereas DYN-NSL-HLE-TSK is rather better. After $T_{1}$ is scheduled on $P_{1}, \operatorname{sl}\left(T_{3}\right)=42, f t\left(P_{1}\right)=12$ and $f t\left(P_{2}\right)=0$ :

$$
\begin{align*}
& d l\left(T_{3}, P_{1}\right)=42-\max (12,12)=30  \tag{10-4}\\
& d l\left(T_{3}, P_{2}\right)=42-\max (12,14)=28 \tag{10-5}
\end{align*}
$$

[^48]ensuring that $T_{3}$ is also scheduled on $P_{1}$ as illustrated in Figure 10-4. From a performance perspective, it is apparent that $O(n m)$ dynamic levels will be created by this procedure.


Figure 10-4: DYN-NSL-HLE-TSK schedule for 4-stage EQ section on a 2-DSP M-P.

Now $\operatorname{sl}\left(T_{5}\right)=27, f t\left(P_{1}\right)=27$ and $f t\left(P_{2}\right)=0$ giving $d l\left(T_{5}, P_{1}\right)=0$ and $d l\left(T_{5}, P_{2}\right)=-2$, so that $T_{5}$ is also scheduled on $P_{1}$. Finally, $T_{7}$ is allocated onto $P_{1}$, minimising $\omega_{s c h}$ and avoiding thrashing. DYN-NSL-HLE-ALL gives the same schedule as above, but not without significant run-time uplift as $O\left(\mathrm{~nm}^{2}\right)$ levels must be generated.

### 10.2 Performance with FWE

In the rest of this chapter the aim is to determine which of the dynamic labelling schemes, detailed in Section 10.1.1, minimises the anomalous behaviour of CPM-xxx-HLE. Here, results are analysed for instances where all IPC is fully overlapped with task execution - FWE.

### 10.2.1 Algorithm Run-Time

DMC taskforces may vary from bus- to tree-based structures with a high degree of inherent parallelism. Consequently, $D C S$ run-time is examined at both extremes of this spectrum.

### 10.2.1.1 Bus-Configured Taskforces

With FWE IPC mechanisms, a rather interesting pattern emerges when algorithm run-time $t$ is plotted as Figure 10-5. PRO and ALL labelling schemes are almost identical with $t>3: 00.00$. dyn_gen_lvi (...) must generate $O\left(m^{2}\right)$ dynamic levels $d l\left(T_{r}, P_{a}\right)$ for PRO and $O\left(\mathrm{~nm}^{2}\right)$ for ALL. When $m » n$, as is the case here, these run-time complexities are effectively equal.

By definition, only one $T_{r}$ is considered in NON and TSK schemes, reducing algorithm run-time to $O(m)$ and $O(n m)$ respectively. Re-plotting Figure $10-5$ on $\log -\log$ axes


Figure 10-5: DYN-FWE-HLE run-time for bus-type taskforces on a 16-DSP M-P.
demonstrates the relationship previously noted in Section 9.4.1. Linear regression gives runtime for the TSK dynamic strategy as $\log t_{\text {TSK }}=2.597 \log c-8.597$ with $\rho=0.9992$.


Figure 10-6: DYN-FWE-HLE $\log t$ for bus-type taskforces on a 16-DSP M-P.

### 10.2.1.2 Tree-Configured Taskforces

With tree-configured mix-buses, NON and TSK labelling schemes prove to be efficient methods for producing DMC schedules, giving run-times in the order of seconds. Although

PRO will generate $O(m)$ ready-task, available-processor pairs for the available processor $P_{a}$ with lowest index at each of $O(m)$ scheduling steps, run-times are still under 1:00.00.


Figure 10-7: DYN-FWE-HLE run-time for tree-type taskforces on a 16-DSP M-P.

This behaviour of PRO is a substantial improvement over the results reported in the previous section. Since all ( $T_{r}, P_{a}$ ) are generated in ALL, this scheme rapidly becomes $O\left(\mathrm{~nm}^{2}\right)$ with run-times in excess of 8:00.00: over 8-times longer than PRO. Replotting results on $\log -\log$ axes as shown in Figure 10-8 gives almost 'linear' characteristics for all 4 schemes.


Figure 10-8: DYN-FWE-HLE $\log t$ for tree-type taskforces on a 16-DSP M-P.

### 10.2.2 Schedule Length

Comparing the $\omega_{s c h} / \omega_{c p}$ ratio for console taskforces with varying degrees of taskforce parallelism, as shown in Figure 10-9, confirms that there is nothing to choose between the schemes until $\pi>1.5$. As $\pi$ increases further, NON and PRO consistently give the shortest $\omega_{s c h}$ with schedules produced by ALL at least $9.29 \%$ longer. Interestingly, the TSK scheme switches between the NON/PRO and ALL characteristics as $\pi$ increases beyond 1.5.


Figure 10-9: DYN-FWE-HLE $\omega_{s c h}$ for tree-type taskforces on a 16-DSP M-P.

### 10.2.3 Speed-Up

While different static labelling schemes yield varying performance as regards taskforce speedup (see Section 9.4.4), Figure 10-10 contains the speed-up characteristics for the four dynamic labelling schemes developed earlier in this thesis. For fully-overlapped (FWE) IPC, all schemes exhibit the same speed-up behaviour such that speed-up is ultimately limited by $\omega_{c p}$ when $n \geq 40$ - for the 64 -channel tree-configured DMC taskforce considered here.

### 10.2.4 Scaled Speed-Up

Similarly, different static labelling schemes exhibit varying scaled speed-up performance. As Figure 10-11 shows however, all four dynamic strategies exhibit identical linear, though subunitary, scaled speed-up. Note that Figure $10-10$ intersects the curve shown below when


Figure 10-10: DYN-FWE-HLE speed-up for tree-type taskforces.
$n=1$ and $n=m$ as predicted in the theoretical work presented in Section 7.2.4. This behaviour is rightly attributed to the ratio of parallel to sequential work increasing with $n$.


Figure 10-11: DYN-FWE-HLE scaled speed-up for tree-type taskforces.

### 10.3 Performance with NSL

The performance of a single non-overlapped IPC channel has been studied from a theoretical point of view in Section 7.4.2. In this section, the performance of dynamic labelling schemes is investigated with a individual IPC channel which supports sequential inter-task messaging.

### 10.3.1 Algorithm Run-Time

With wide variation in run-time characteristics already reported for CPM-FWE-xxx and DYN-FWE-HLE-xxx, here $t$ is investigated for NSL IPC mechanisms with the same DMC taskforces.

### 10.3.1.1 Bus-Configured Taskforces

$D C S$ run-time for the four dynamic-labelling schemes using NSL is plotted in Figure 10-12. While similar to the results of Section 10.2.1, it is apparent that TSK is preferred with NSL IPC. Although NON gives comparable performance, that TSK selects the first $T_{r}$ and then generates all $\left(T_{r}, P_{a}\right)$ pairs ensures that $T_{r}$ is scheduled on that $P_{a}$ such that $d l\left(T_{r}, P_{a}\right)$ is minimal.


Figure 10-12: DYN-NSL-HLE run-time for bus-type taskforces on a 16-DSP M-P.

Incorporating IPC costs into the $D C S$ algorithms evidently improves the choice of processor on which to schedule a given task. The same reasoning also explains why ALL
exhibits similar run-time characteristics for NSL as for FWE IPC detailed in Figure 10-5. Under NSL, PRO is clearly the worst scheme giving run-times well in excess of 5:00.00.

### 10.3.1.2 Tree-Configured Taskforces

With tree-type console configurations, run-time characteristics are obtained as illustrated in Figure 10-13. These results are almost identical to those presented for FWE IPC mechanisms, except that ALL run-times are reduced by a factor of 0.560 . When considered from a $\log -\log$ perspective, the NSL scheme yields: $\log t_{\text {TSK }}=2.429 \log c-8.240$ with $\rho=0.9994$, which bears close resemblance to that given previously for fully-overlapping IPC channels.


Figure 10-13: DYN-NSL-HLE run-time for tree-type taskforces on a 16-DSP M-P.

### 10.3.2 Schedule Length

When NSL is considered via a $\omega_{s c h} / \omega_{c p}$ comparison as Figure $10-14$, it is clear that IPC directly effects $\omega_{\text {sch }}$ throughout the entire range of $\pi$. Not surprisingly, NON is the worst scheme due to a lack of consideration of the current state of tasks and processors. True dynamic strategies give rather better performance: PRO, TSK and ALL intersect at several points, though TSK and ALL generally give better results for this pessimistic IPC mechanism.


Figure 10-14: DYN-NSL-HLE $\omega_{s c h}$ for tree-type taskforces on a 16-DSP M-P.

### 10.3.3 Speed-Up

As Figure $10-15$ shows, an unfamiliar pattern emerges when NSL speed-up is plotted for dynamic labelling. Both NON and PRO ultimately converge to a value some $27.04 \%$ less than the maximum speed-up determined by $\omega_{c p}$. ALL outstrips all other schemes until it converges at $n=32$, except for a value of almost $10.00 \%$ less than $\omega_{s e q} / \omega_{c p}$ for $n=24$. In terms of consistent performance, however, TSK is the best strategy to maximise taskforce speed-up.


Figure 10-15: DYN-NSL-HLE speed-up for tree-type taskforces.

### 10.3.4 Scaled Speed-Up

Scaling $c$ as $n$ increases gives the characteristics displayed in Figure 10-16. Both TSK and ALL give results which adhere closely to the theory developed in Section 7.2.4, though TSK never quite converges to the conventional speed-up of 3.24 which ALL reaches when $n=64$. NON and PRO are very poor from this perspective since NON converges to a value only $57.43 \%$ of that achieved by ALL, whereas PRO actually moves away from this asymptote for $n \geq 32$.


Figure 10-16: DYN-NSL-HLE scaled speed-up for tree-type taskforces.

### 10.4 Performance with NML

One decisive assumption in NSL is that all IPC proceeds sequentially. It is perfectly possible, as developed for HYMIPS in Chapters $5 \& 6$, to replicate architectural features for IPC. Hence, results are presented for DYN scheduling using multiple non-overlapped IPC channels.

### 10.4.1 Algorithm Run-Time

As the results in Figure 10-17 show, the run-time performance of dynamic labelling using bus-configured taskforces with NML is virtually identical to that obtained with NSL IPC. When tree-type taskforces are scheduled with NML, Figure 10-18 illustrates that algorithm run-time is again identical to NSL mechanisms. This behaviour can be explained in part by the fact that while reserving IPC is an integral part of DYN, it is ignored in the static labelling process.


Figure 10-17: DYN-NML-HLE run-time for bus-type taskforces on a 16-DSP M-P.


Figure 10-18: DYN-NML-HLE run-time for tree-type taskforces on a 16-DSP M-P.

### 10.4.2 Schedule Length

Plotting the schedule length to critical path ratio, $\omega_{s c h} / \omega_{c p}$, against normalised taskforce parallelism, $\pi$, gives the graphs for NML included in Figure 10-19. Again the linear trend is apparent especially in NON and PRO labelling schemes. PRO is consistently good at minimising
the $\omega$ ratio for the range of $\pi$ considered here. The performance of TSK and ALL proves to be rather erratic: lying below PRO in a few instances but lying on or above NON for others.


Figure 10-19: DYN-NML-HLE $\omega_{s c h}$ for bus-type taskforces on a 16-DSP M-P.

### 10.4.3 Speed-Up

Graphing speed-up for NML gives characteristics not previously seen in this analysis. TSK is precisely linear sub-unitary until speed-up is constrained to 7.41 by $\omega_{c p}$. NON follows TSK for small $n$, eventually converging to 6.00 . This lower value is also met by PRO, which seems to be the poorest labelling scheme as regards speed-up for NML IPC. ALL fluctuates in a similar fashion to TSK, but ultimately converges to the same maximum speed-up as TSK.

### 10.4.4 Scaled Speed-Up

Figure 10-21 details the scaled speed-up of dynamic strategies with NML. Since no re-labelling is undertaken in NON, the state of the partial schedule is not considered and so gives worst performance. PRO fluctuates about a generally linear trend as $n$ increases, but still lies some $35.52 \%$ beneath TSK and ALL when $n=64$. TSK and ALL arrange IPC overhead in such a way as to give identical linear SSU for the range of $\mathrm{M}-\mathrm{P}$ architectures considered here.


Figure 10-20: DYN-NML-HLE speed-up for bus-type taskforces.


Figure 10-21: DYN-NML-HLE scaled speed-up for bus-type taskforces.

### 10.5 Task Granularity

M-P system architects have long debated the relative merits of fine and coarse granularity. Accordingly, the performance of DYN-NML-HLE-TSK is measured using DMC taskforces exhibiting a range of task granularity, or $r / c$, as previously defined in Section 7.4.1.

### 10.5.1 DMC Implementation

The theoretical investigation in Chapter 7 has shown that DMC depends strongly on $r / c$. As Figure 10-22 illustrates, $r / c$ is high in coarse-grained parallelism and IPC overhead can be overlapped during many instruction cycles. Conversely, in fine-grained taskforces the overhead per unit computation $r / c$ is relatively low. Many more PEs may be efficiently utilised than in coarse-grained schemes, but not without significant uplift in IPC overhead.


```
Function: MIX_BUS
    fan in:l mix-bus primitive
    as per Figure 9-2
Resource Requirements:
    instruction cycles - (10 + fan_in) cycles
    P memory - 0 words
    X-data memory - 0 words
    Y-date memory - 0 words
MIX BUS \
        initialise pointer and modulo registers
        org p:
        move #inputs,r0
        move #fan_in,mo
        move #params,r4
        move #fan_in,m4
        perform fan_in:1 mix-bus kexmel
```



```
        clr a
        move }x:(x0)+,x0\quady:(r4)+,y
        .LOOP #fan_in-1 mac mo,y0,a 
        ENDL
        macr x0,yo,a
        move a,x:output
    ENDM ; MIX buS
```

Figure 10-22: DSP56001 assembly code for generic DMC mix-bus kernel. ${ }^{\text {a }}$
a. Note that audio rate interpolation of control rate data is external to this macro.

The reader's attention is drawn to the allocation of a 16 -input DMCs onto a 8-PE M-P detailed in Table 10-1. Dynamics ASP was removed from four of the sixteen channels, allowing this 16:8 taskforce to meet the constraints imposed by $f_{s}$ [Linton, 1991]. Reducing $m$ and $\omega_{s e q}$ together by replacing the 2 -input mix tasks, $e()=12^{1}$, with 16 -input primitives, $e()=26$, can significantly reduce the $D C S$ algorithm run-time scheduling overhead.

[^49]| Console <br> Taskforce | Sequential <br> Schedule | Parallei <br> Schedule | Algorithm <br> Run-time | Speed-up | Processor <br> Efficiency |
| :---: | :---: | :---: | :---: | :---: | :---: |
| $16: 1$ | 2044 | 268 | $0: 07.57$ | 7.62 | $95.31 \%$ |
| $16: 2$ | 2104 | 268 | $0: 11.65$ | 7.85 | $98.13 \%$ |
| $16: 4$ | 2224 | 280 | $0: 23.28$ | 7.94 | $99.30 \%$ |
| $16: 8^{\mathrm{a}}$ | 2214 | 278 | $1: 04.83$ | 7.96 | $99.48 \%$ |

Table 10-1: DYN-NML-HLE-TSK schedules for various 16 -input consoles on an 8-DSP M-P.
a. Dynamics processing was removed from four channels.

### 10.5.2 Algorithm Run-Time

Variation in run-time is anticipated due to the number of primitives required to construct a mix-bus. For example, it $i$ is the number of inputs supported and $b$ represents the fan-in of each mix primitive (see Figure 9-2), then the number of mix tasks required $n_{m}$ is given by: ${ }^{1}$

$$
\begin{equation*}
n_{m}=\left\lceil\frac{i-1}{b-1}\right\rceil \tag{10-6}
\end{equation*}
$$

### 10.5.2.1 Bus-Configured Taskforces

As illustrated in Figure 10-23, there is clear distinction in run-time $t$ when scheduling 64-channel bus-configured consoles constructed from mix primitives of varying granularity. While the 16 -input console requires less than $00: 01.00$ to be scheduled on the 32-DSP topology considered here, almost a minute is required if the taskforce uses 2-input primitives.

Re-plotting these experimental results on logarithmic axes as shown below in Figure 10-24 reveals that all curves are linear in this domain. Performing linear regression on these values gives $\log t_{2}=2.394 \log c-7.710$ and $\log t_{16}=0.785 \log c-7.091$ for 2-input and 16-input primitives, respectively, with product-moment correlation $>0.9878$.

### 10.5.2.2 Tree-Configured Taskforces

Repeating this analysis using taskforces configured with mix-trees results in similar characteristics, as Figure 10-25 illustrates. Run-time for 64 -channels with 2-input primitives is some $21.78 \%$ above that reported in the previous section. Evaluating these results from a

[^50]

Figure 10-23: DYN-NML-HLE-TSK run-time for bus-type taskforces on a 32-DSP M-P.


Figure 10-24: DYN-NML-HLE-TSK $\log t$ for bus-type taskforces on a 32-DSP M-P.
logarithmic perspective gives $\log t_{2}=2.286 \log c-6.728$ and $\log t_{16}=1.610 \log c-7.522$, with Pearson product-moment correlation $>0.9918$ in both instances.

### 10.5.3 Schedule Length

Calculating $\omega_{s c h} / \omega_{c p}$ allows these alternative DMC construction techniques to be compared for various values of $\pi$. From Figure $10-26$, it is evident that employing 16 -input primitives is


Figure 10-25: DYN-NML-HLE-TSK run-time for tree-type taskforces on a 32-DSP M-P.
the best method of minimising the $\omega$ ratio. The results presented in this illustration would tend to suggest that 2 -input tasks tend to be more effective than 4 - or 8 -input alternatives.

Increasing $\pi$ further, however, shows that this is not the case and the ordering is -16 , $8,4,2$. It must be remembered that when $\pi=4.00, \omega_{c p}=18$ and $\omega_{s c h} / \omega_{c p}=1.40$ for 2-input implementations whereas $\omega_{c p}=36$ and $\omega_{s c h} / \omega_{c p}=1.33$ with 16 -input tasks. In other words, the 2 -input schedule is $47.50 \%$ less than that obtained with 16 -input primitives.


Figure 10-26: DYN-NML-HLE-TSK $\omega_{s c h}$ for tree-type taskforces on a 32-DSP M-P.

### 10.5.4 Speed-Up

It is evident from Figure 10-27 that 2-input tasks return the best speed-up performance. While the 2 -input curve is close to linear until $\omega_{s c h}$ reaches $\omega_{c p}$ when $n=48$, this occurs with 4input primitives when $n=24$. Similar behaviour is observed for the other cases, except that 8 -input converges to the $\omega_{c p}$-constrained value when $n=12$. The 16 -input curve converges when $n$ is as low as 8 , such that speed-up is only $28.88 \%$ of the 2 -input case when $n=64$.


Figure 10-27: DYN-NML-HLE-TSK speed-up for tree-type taskforces.

### 10.5.5 Scaled Speed-Up

When scaled speed-up (SSU) is the performance metric similar behaviour is observed. All four alternatives are close to linear throughout, with the 2 -input characteristic as high as 7.41 when $n=64$. Using this size of DMC and M-P, the 4 -input SSU is $56.67 \%$ of this value, with 8 - and 16 -input primitives yielding $35.71 \%$ and $28.91 \%$ respectively. At this point, the SSU curves intersect with the corresponding conventional speed-up characteristics.


Figure 10-28: DYN-NML-HLE-TSK scaled speed-up for tree-type taskforces.

### 10.6 Conclusions

In DMC scheduling, the central aim is to efficiently allocate a DMC taskforce with $m$ ASP tasks onto a $n$-processor M-P architecture. Worst-case $O\left(n^{m}\right)$ trends in AST and anomalies in CPM motivate further research into list-scheduling. MIA and MRC heuristics studied previously, suggest that HLE labels should be revised within the DMC scheduling process.

Equation (10-1) combines aspects of both preferred AST heuristics - MIA and MRC in order to assign highest priority to that ( $T_{r}, P_{a}$ ) incurring minimum extension of the current partial schedule. Three alternative schemes for task and processor selection - PRO, TSK and ALL - are defined and an upper bound derived for the dynamic strategies supported by DCS. ${ }^{1}$

Scheduling an EQ section from Figure 9-1 immediately illustrates some of the advantages engendered by the dynamic approach to DMC taskforce labelling. Scheduling the problem instance shown in Figure $10-4$ with DYN-NSL-HLE-TSK not only reduces $\omega_{\text {sch }}$ to some $90.00 \%$ of the DYN-NSL-HLE-NON case, but processor thrashing is also eliminated.

An noted in Section 10.2, the primary objective in this chapter is to determine which, if any, dynamic-labelling scheme is more apt for DMC task scheduling than CPM-xxx-HLE. Rather than extract general trends with randomly-generated taskforces, console designs

[^51]entered via schematic capture are used throughout this analysis. With fully-overlapped (FWE) IPC mechanisms, $D C S$ run-time for PRO and ALL with bus-configurations is almost identical.

Referring to Section 10.2.1, dyn_gen_lvl (...) will generate $O\left(m^{2}\right)$ dynamic levels for PRO and $O\left(\mathrm{~nm}^{2}\right)$ for ALL. When $m » n$, as is typically the case in the DMC problem domain, PRO and ALL complexities tend to the same value in the limit. Employing tree-type taskforces shows that NON and TSK are particularly efficient strategies for generating DMC schedules. Run-time characteristics are effectively linear in the logarithmic domain.

In terms of $\omega_{s c h}$, NON and PRO appear to be the best techniques for revising task labels, though TSK gives similar values in many instances. All dynamic schemes exhibit identical speed-up and SSU performance under FWE, intersecting when $n=1$ and $n=m$ as predicated in Section 7.2.4. While this IPC transport model is perhaps over-optimistic, HYMIPS has shown that it is in fact possible to fully overlap computation with IPC.

Repeating this analysis with NSL media shows that $D C S$ run-time for bus-configured taskforces is extremely close to that with FWE. While NON yields performance comparable to TSK, it is evident that the choice of $P_{a}$ on which to schedule $T_{r}$ is improved by dynamiclabelling when IPC messaging modifies the partial schedule. Selecting the available processor first, PRO, gives run-times well in excess of 5:00.00 for the 64-channel console used here.

With tree-configured taskforces, run-time with NSL is identical to that for FWE transport, except that $t_{\mathrm{ALL}}$ is reduced by a factor of 0.560 . From Figure $10-14$, it is evident that IPC directly affects $\omega_{s c h} / \omega_{c p}$ throughout the entire range of $\pi$. NON is the worst scheme studied here due to the current task- and processor-list being ignored. Fortunately, TSK and ALL yield rather better results for this somewhat conservative IPC mechanism.

Figure 10-15 reveals that NSL speed-up is markedly different when compared with FWE. Both NON and PRO converge to a value some $27.04 \%$ less than the maximum level determined by $\omega_{c p}$. Both TSK and ALL SSU is no longer linear due to the impact of nonoverlapped IPC. PRO is very poor in this regard with PRO $\frac{\partial}{\partial n}$ actually negative for $n>32$.

While it may not be possible to overlap inter-task transport with computation in some audio M-Ps, it is quite feasible to replicate architectural features. With multiple NSL channels
(NML), DCS run-time for bus-configurations is almost identical to FWE. Tree-type DMCs require run-times similar to NSL, as reserving IPC is an integral part of the DYN algorithm.

Linear $\omega_{s c h} / \omega_{c p}$ behaviour is readily apparent for NON and PRO labelling schemes under NML. In fact, PRO is consistently able to minimise the $\omega$ ratio for the range of $\pi$ considered in this analysis. Unfortunately, the performance of TSK and ALL dynamic-labelling schemes proves to be rather erratic, but does lie below PRO in a few problem instances.

Charting conventional speed-up under NML gives characteristics not previously seen in this analysis. TSK is precisely linear sub-unitary until constrained by $\omega_{c p}$, whereas NON gives very variable results with PRO consistently poor. TSK and ALL arrange IPC in such a way as to give identical linear SSU for the range of DMC configurations considered here.

Performance diversity is anticipated when comparing taskforce granularity using DYN-NML-HLE-TSK as $n_{m}$ varies as Equation (10-6). $D C S$ confirms that run-time with larger mix-bus primitives is considerably less than for 2-input variants. Tree-configurations extend $t$ by up to $21.78 \%$ due to the increased number of $T_{r}$ at any step in the DYN scheduling process.

16-input primitives appear to be the best method of minimising $\omega_{s c h} / \omega_{c p}$. However, parallelism inherent in DMCs means that $\omega_{\text {sch }}$ for 2-input primitives is $47.50 \%$ less than that obtained with 16 -input tasks. This is due to $\omega_{c p}$ in consoles constructed from 16-input mixbus tasks being greater than that evident in comparable 2-input implementations.

Due to the increase in task size, large-grain mix-bus primitives exhibit poor speed-up performance. $S(n)$ quickly converges to the $\omega_{c p}$-constrained value, since the opportunities for $n$-fold parallelism are severely restricted. All four mix-bus structures return linear SSU characteristics, with larger fan-in giving lower SSU as explained for conventional speed-up.

### 10.7 References

[Graham, 1976] Graham R L: "Bounds on the Performance of Scheduling Algorithms", from Computer and Job-Shop Scheduling Theory, Wiley \& Sons, 1976, ISBN 0-471-16319-8. [Linton, 1991] Linton K N (et al.): "On the Re-Allocation of Processing Resources for DASP", IEE Colloquium on Digital Audio Signal Processing, London, UK, May 1991.

## Chapter 11

## Conclusions \& Further Work

Commercial and creative pressures stimulate considerable interest in the MIMD paradigm for professional digital mixing. This thesis focuses on salient issues in parallel architectures and task scheduling strategies. Herein the author provides new insight into the major hardware and software problems which must be resolved prior to the development of such products.

### 11.1 Digital Mixing Consoles

A number of factors influence DMC design, ranging from control ergonomics and automation systems through to real-time ASP and underlying M-P architecture. MMI and TDA opportunities are among the most compelling reasons to favour a fully-digital methodology.

### 11.1.1 Analogue Mixing Consoles

ASP has defined the recorded sound of each musical era, with spatialisation systems stressed today. Although professional consoles contain considerable processing facilities, including basic control of spatial parameters, more comprehensive ASP is still outboard. DSP has added to the repertoire of effects, taking full advantage of decades of evolution in unrelated fields.

Console size and component density are constant design problems if performance levels are to be retained as analogue design endeavours to meet operational requirements. Rather than developments in processing technology, the author predicts that advances in professional console system technology will define recorded sound in the coming decade.

While some digital consoles exist in the current marketplace, none provide either the degree or type of functionality originally anticipated by industry professionals. Control surface and automation software are especially critical in this regard. It is therefore essential to investigate DMC implementation issues as they affect both console designer and operator.

### 11.1.2 Console Digitalisation

Recent trends, including SP-DIF and MADI, augur digital implementation of the sound production process between microphone capsule and loudspeaker cone. A vital stage in DMC design is the ASP sub-system itself. Investigating this aspect not only determines the real-time processing budget, but also reveals redundancy inherent during tracking and mix-down.

Much effort has been invested in advancing digital ASP techniques. In contrast, control surface layout is not generally perceived as important. Freedom from the constraints imposed by mechanical linkage creates a number of opportunities to improve the MMI. Digital implementation makes TDA a technical reality although at significant cost uplift.

Alas, it would seem that the MMI is becoming more maladroit with each advance in surface technology. Improvements can, however, be achieved solely as a result of surface design and not advanced technology. As the heart of any console design must be the ASP engine itself, the second part of this thesis concentrates on this critical aspect of DMC desigu.

### 11.2 Parallel Architectures

Programmable DSP devices and architectural issues are investigated from a digital mixing perspective. Alternative topologies are compared as they directly effect DMC performance, modularity and extensibility. Design decisions are justified for a hybrid audio M-P.

### 11.2.1 Programmable DSP Devices

Floating-point devices largely free the DMC designer from concerns about scaling. Processors such as Motorola's DSP core and the TASP family from Texas Instruments provide stiff competition for ASICs. Although exotic VLSI may achieve another order-of-magnitude improvement in clock rate, industry expectations will undoubtedly match any such advance.

Given stringent VLSI packaging restrictions, byte-wide IPC links provide efficient inter-processor connectivity. While these developments support M-Ps, few devices integrate specific features for DMC systems directly on-chip. None support dedicated delay processing or the additional modulation waveforms required for chorus and pitch-shift algorithms.

In order to implement the console mix function, many instruction cycles are consumed interpolating control-rate data. Multimedia parts such as the Philips Trimedia and SGSThompson A110 are emerging with specific DSP features targeted at audio applications. Cost compares favourably with the original pricing structure of the Motorola DSP56001.

### 11.2.2 Architectural Issues

An original schema of topology metrics, implemented in the $T P G$ deliverable, enables quantitative comparison of audio console M-Ps. With a link expansion factor of one, the shared-bus is well-suited to small-scale DMCs until contention limits extensibility. Ring architectures, particularly the tri-valent chordal ring, deserve special attention from the DMC designer. Cross-topology links reduce the maximum inter-node distance $\delta$ by a factor of two.

Expanding tree-based DMCs via fan-out $b$ cannot be recommended, whereas incrementing $l$ maintains a fixed number of IPC ports. Less familiar topologies such as cubeconnected cycles are among the most attractive constant-valency M-Ps. An imbalance of inter-node traffic is apparent near the root node of a simply $b$-ary tree. Ring connections in the $x$-tree ease this bottle-neck while still matching closely in-tree precedence constraints.

If a range of mixing products is the manufacturer's primary goal, trees with restricted fan-out enable a structured upgrade path. At least 3 links must fail in the chordal-ring, x-tree, and cube-connected cycles before a PE is isolated. Unfortunately, the shared-bus is least suitable from this point of view. Chordal-rings and tree structures must be the preferred topologies for DMC implementation, though multi-stage adaptations should not be precluded.

### 11.2.3 Audio M-P Implementation

Latency in message-passing schemes is dependent on the channel bandwidth, distance and message size. Wormhole routing greatly reduces IPC latency in DMCs as delay is no longer sensitive to the distance involved in sample transport. The IPC latency in a 2-D mesh may be reduced up to $O(\sqrt{n})$ times over a binary hypercube if constant bisection bandwidth can be maintained. Fewer links contribute to the bisection, permitting each channel to be made wider.

Assuming constant PE bandwidth, IPC latency can be reduced by a up to a factor of 2 if a higher-dimensional $n$-processor hypercube is re-configured as a 2-D mesh. X-tree based

DMCs do not perform especially well under these conditions since 2 connections are required per processor to interconnect the ring at each level. A simple 3-ary tree gives the same wormhole latency improvement over a binary hypercube as a chordal-ring when $k$ « $K$.

HYMIPS is not a shared-bus M-P in the classical sense: shared DPR and an alternative semaphore protocol allow pseudo-wormhole messaging to overlap task execution, supporting a range of virtual IPC topologies. A HYMIPS node has been constructed, together with HSR and HST SP-DIF peripherals. This hybrid approach to the DMC compute engine offers scalability both in terms of the number of compute PEs per node and the number of nodes.

### 11.3 Task Scheduling Strategies

In DMC scheduling, the central aim is to efficiently allocate a taskforce with $m$ ASP tasks onto a $n$-processor M-P. Worst-case trends in AST and anomalies in CPM have instigated research into list-scheduling. Dynamic-labelling captures the significant features of DCS.

### 11.3.1 Multi-Processor Models

DCS must observe all precedence relations and be non-preemptive. Since parameters are known a priori, the DMC taskforce can be scheduled at compile-time. Conventional speed-up assumes that the ratio of parallel to sequential work is fixed. In situations were both taskforce and M-P are scaled, SSU indicates how performance will vary as the entire console is scaled.

As noted in Chapter 3, the maximum number of pipeline stages in a LPP-based DMC is $k_{\max }=48$. Further analysis shows that this approach to parallelising the DMC taskforce provides opportunities to achieve unitary speed-up and maximum processing efficiency. This technique has been directly applied within the HYMIPS audio M-P described previously.

A number of different M-P models have been investigated with respect to task granularity. Unitary speed-up can be achieved by overlapping computation with multiple IPC links. For each model, there is some maximum $n$ that is cost-effective: this number depends on the M-P topology, on IPC mechanisms and, to a degree, on the particular DMC taskforce.

### 11.3.2 Scheduling Strategies

DMC DAGs have task execution times known a priori. With the complexity of the DCS problem derived in Equation (8-1), the borderline between polynomial and $N P$-complete problems is evident. Important cases such as unit-length tasks with arbitrary precedence for $n \geq 3$, and taskforces with in-tree precedence constraints have particular relevance to DCS.

Both SA and GA require empirical tuning of implementation-specific factors to guarantee convergence. Depending on cooling schedules or recombination operators are not options in the DMC environment. Heuristic underestimates - MRC, MIA and MHU - and static-labelling schemes - HLN, HLE, LCN and LCE - leverage off precedence constraints.

The best priority lists are those in which tasks on longest $\omega_{c p}$ appear near the head. Only relatively small additions to processor usage are then made at the end of the schedule. Lack of idle time and non-preemption are responsible for the umpredictable behaviour of CPM, when available processors are forced to execute ready tasks prematurely.

### 11.3.3 DCS Research Framework

Instead of requiring partitioning a priori, LGDF console schematics are inherently partitioned at the ASP task. Behaviour models test AST, CPM and DYN algorithms, as well as validate NSL, NML and FWE IPC mechanisms. Internally, $D C S$ retains the current state of the partial schedule together with a list of unscheduled tasks. By referring to the Prolog source, the scheduler predicate hierarchy illustrates the benefits of a declarative approach to DCS.

While MPC may be effective as $D C S$ nears completion, this underestimate readily causes AST to revert to exhaustive search. Rather better results are reported for MRC and MHU although the latter requires significant run-time uplift as three separate $h(n)$ must be determined for each $n$. The essentially $O\left(n^{m}\right)$ requirements of AST moves the focus of this research towards deterministic approaches derived from MRC \& MIA heuristic underestimates.

With linear taskforces, no improvement is possible by ordering the priority list, whereas HLE returns manifest benefits when task priority is not fully constrained. With HLE consistently returning shortest $\omega_{s c h}$, $\pi$ has a direct bearing on $\omega_{s c h}: n<\pi$ signifies that
inherent parallelism cannot be fully exploited. In summary, the evidence is clear that the order of non-increasing performance of static-labelling schemes is - HLE, LCN, LCE, HLN, RND.

### 11.3.4 Dynamic Labelling Schemes

Dynamic labelling combines aspects of both preferred AST heuristics in order to assign highest priority to that ( $T_{r}, P_{a}$ ) incurring minimum extension of the current partial schedule. Three alternative schemes - PRO, TSK and ALL - are employed in the DCS deliverable. Scheduling an EQ section in isolation immediately illustrates some of the advantages engendered by dynamic taskforce labelling since 'processor thrashing' is eliminated.

Fully-overlapped (FWE) IPC demonstrates that NON and TSK are efficient strategies for generating DMC schedules. Analysis with NSL media clearly shows that the choice of $P_{a}$ on which to schedule $T_{r}$ is improved by dynamic-labelling. $D C S$ run-time with NML is similar to NSL, as reserving IPC is an integral part of DYN. TSK and ALL arrange IPC in such a way as to give the following ordering, since ALL is $O(m)$ greater than TSK- TSK, ALL, PRO, NON.
$D C S$ confirms the expected performance diversity when comparing taskforce granularity using DYN-NML-HLE-TSK - the preferred approach to DMC scheduling. Inherent parallelism means that $\omega_{\text {sch }}$ for small-grain primitives is considerably less than that obtained with large-grain tasks. Large-grain mix-buses exhibit poor speed-up performance as $n$-fold concurrency is severely restricted. All mix-tree structures return linear SSU characteristics.

### 11.4 Further Work

As the audio community becomes accustomed to buying more technology with less money, manufacturers face increasingly difficult design problems. Concepts from closely related academic subjects are becoming increasingly pertinent in this interdisciplinary industry.

### 11.4.1 Digital Mixing Consoles

With the availability of 'true' DMCs, the author is convinced that proprietary outboard ASP, and spatialisation algorithms in particular, will migrate directly into the console channel strip.

Using the DISQ M-P, AT\&T has demonstrated the use of a known analogue MMI (an SSL $G$-Series) to drive a prototype DMC core, implying interchangeable MMIs in the future.

Automation systems have developed to the extent that many of the final creative decisions can be left until the last possible moment. With the realisation of TDA, the step from manual tracking to computerised mix-down will be seamless. The author perceives that this type of 'tapeless studio' will stimulate fresh ideas in contemporary music production.

### 11.4.2 Parallel Architectures

In this thesis, the author has clearly the demonstrated the viability of constructing audio M-Ps from off-the-shelf DSP parts. As the HYMIPS principle is not unduly restrictive, two further areas of research are implied. Firstly, rather than a dedicated communications processor, ASIC technology presents a cost effective method for message-passing between compute PEs.

Note that the HYMIPS backplane has a simple static RAM interface which connects transparently to most programmable DSP designs, irrespective of RWM support. Secondly, with the unit-cost of new DSP devices dropping to acceptable levels, HYMIPS I would utilise these novel DSP processor within the same generic structure employed in the original design.

### 11.4.3 Task Scheduling Strategies

Whether for development or operational purposes, task scheduling strategies are crucial to the realisation of flexible DMC systems for the professional audio mixing environment.

### 11.4.3.1 Additional Precedence Constraints

Bussel (et al.) [Bussel, 1974] address the general problem of minimising the number of processors, $n$, or the execution time. Reduced to the simplest terms, their algorithm consists of adding precedence constraints (i.e., additional nets) to the original DAG at those points where the number of ready tasks exceeds $n$, or in the terminology developed here, $\pi>1$.

The effect is to distribute processor demands throughout the length of the graph, without adversely affecting overall $\omega_{s c h}$. One technique to add constraints in $D C S$ would be to first schedule the DMC taskforce in reverse using LCN or LCE. Information derived from such schedules could then be used to determine where additional arcs may be added.

### 11.4.3.2 Bus-Configurations Re-Examined

In the results presented for $D C S$ algorithms, it is evident that bus-configured DMCs quickly exceed the pre-schedule $\omega_{c p}$ test detailed in Section 9.2.4. However, the LPP concept developed in Section 7.3 shows one solution to this problem. Careful insertion of pipeline delays within $D C S$ would ensure that linear taskforces directly benefit from this approach.

Alternatively, there is another way to manage DMC taskforces constructed from mixbus primitives. With interpolated control-rate fader data and audio rate sample data adhering to associative principles, provided extreme numerical effects such as overflow are absent, then any mix-bus structure, $\omega_{c p}=\omega_{\text {seq }}$, can be re-configured as a mix-tree $\omega_{c p}$ " $\omega_{s e q}$.

### 11.4.3.3 Code Generation

DSP assembly code macros, see Figures $9-5$ and $10-22$, allow sound software engineering practices to be employed. One consequence is, that in order to maintain modularity, no task can retain ownership of DSP ALU registers. When a task is launched, no assumptions can be made about the data held in these registers. The entire task context must be stored in memory.

Modular DSP coding techniques lead to redundant instruction execution. In the code fragment of Figure 11-1 accumulator $a$ is stored as task $T_{1}$ completes and is immediately reloaded with the same value when $T_{2}$ commences execution. Unnecessary overhead could be removed by post-allocation boundary optimisation and the code file subsequently compiled.


Figure 11-1: Example of wasteful code at DMC task boundary.

### 11.4.4 Manufacturing Considerations

There are many ways to express the performance of a DMC. While the performance metric commonly used is MIPS cost, fault-tolerance and extensibility are just as important.

### 11.4.4.1 Cost

An understanding of cost is essential for designers to be able to make informed decisions. The cost of DSP components is changing so fast that design decisions must not be based on
today's costs, but on projected costs at the time the product is shipped. For example, in digital audio component costs typically make up 15-33\% of the final list price [Scuffham, 1992].

Most companies spend only 8-15\% of income on R\&D, which suggests the application of fixed-overhead percentages to translate cost into price. However, if this level of R\&D is considered as an investment then every pound spent on R\&D must generate $£ 7-13$ in sales. This alternative point of view would then imply different gross margins depending on product.

### 11.4.4.2 Fault-Tolerance

It is commonly held that M-Ps are inherently tolerant of faults since there is a natural availability of spares. Removing a node effectively removes all the connected arcs, with the result that node failure is more 'damaging' than arc failure. Designing DMCs which are faulttolerant (and hence reliable) involves a great deal more that simply providing spares, however.

The only alternative is to design multi-DSP systems for maximum resilience and faulttolerance. Designing for maximum resilience means minimising the use of the least reliable components. Designing for fault-tolerance means two things: firstly, detection of the occurrence of an error, and; secondly, correction and recovery from a detected error.

### 11.4.4.3 Extensibility

Few customers can afford the initial cost of a large multi-DSP audio system. However, many can afford to grow a large system in increments as their budgets permit. There is also a great risk in designing a multi-DSP M-P welded to a particular ASP application. By incorporating sufficient flexibility, $\mathrm{R} \& D$ investment can be amortised over a range of end-products.

It is obviously desirable to minimise the increase in cost to that of linear growth. Moreover, the designer must assess how architectural parameters, such as IPC capacity, vary with replication. An important practical consideration, especially for very large-scale ASP, is the space occupied by the system as a whole and, more particularly, by inter-processor wiring.

### 11.5 Summary

Large DSP systems generally cost more to develop - a product costing 10 times as much to manufacture may cost many times as much to develop. As yet, there is no expedient way to
developing digital mixing products which derive full benefit from digital implementation. $T P G, H Y M P S$ and $D C S$ contribute directly to meeting the demands of audio professionals.

HYMIPS, a hybrid audio M-P targeted at real-time multi-channel ASP, illustrates ways in which programmable DSP parts can be employed to construct real-time ASP engines. TPG compares $12 \mathrm{M}-\mathrm{P}$ topologies using 13 metrics to demonstrate that chordal-ring, tree and cubeconnected cycles are the topologies of choice for architectures underlying DMC systems.

Dynamic-labelling significantly improves performance by over $50.00 \%$ when compared to CPM-xxx-HLE alone. Although the inability of DYN-xxx-HLE-TSK to idle processors is an inherent flaw in any list-scheduling strategy, the evidence seems clear that this particular approximation algorithm is well suited to the professional DMC environment.

### 11.6 References

[Bussell, 1974] Bussel B, Fernandez E and Levy O: "Optimal Scheduling for Homogeneous Multiprocessors", Proc. IFIP Congress 74, September 1974.
[Scuffham, 1992] Scuffham M: JFX Hardware Costing, private communication, November 1992.

## Appendix A

## Survey of DSP Processors

Architecturally important programmable DSP processors are reviewed in Table A-1. It should be noted that many popular devices have subsequently been released with faster clock rates.

| Company | Part No. | Release Date | MAC (ns) | Comments |
| :---: | :---: | :---: | :---: | :---: |
| AMI | s2811 | 1978 | 300 | First DSP design, 12/16-bit FX, released 1982 |
| Analog Devices | ADSP-2 100/00A | 1986/88 | 125/80 | 16/40-bit FX |
|  | ADSP-21020 | 1991 | 80 or 100 | $\begin{aligned} & \text { 32-bit FP, } \\ & 40 \text {-bit IEEE FP, } \\ & 2 \text { external buses } \end{aligned}$ |
| AT\&T | DSP1 | 1979 | 800 | Early DSP, 16/20-bit FX, not marketed |
|  | DSP32/32C | 1984/88 | 160/80 | 32-bit FP |
|  | DSP16/16A | 1987/88 | 55/33 | 16-bit FX |
|  | DSP3210 | 1992 | 30 or 36 | Enhanced DSP32, with host port |
| Fujitsu | MB8764 | 1983 | 120 | 16/26-bit FX |
|  | MB86232 | 1987 | 150 | 32-bit IEEE FP |
|  | MB86224 | 1989 | 75 | 24-bit FX |
| Hitachi | HD61810 | 1982 | 250 | 12-bit FP |
|  | DSPi | 1988 | 50 | Very fast I/O, for image processing |
| IBM | Hermes | 1981 | 100 | Fast MAC time, not marketed |
| Inmos | IMS A100 | 1986 | (12.5) | 16-bit systolic FX, based on 32-tap transversal filter |

Table A-1: Summary of important DSP devices from VLSI manufacturers. ${ }^{\text {a }}$

| Company | Part No. | Release Date | MAC (ns) | Comments |
| :---: | :---: | :---: | :---: | :---: |
| Motorola | DSP56001 | 1987 | 60,75 or 97.5 | 24-bit FX, aimed at digital audio |
|  | DSP96002 | 1990 | 60 or 75 | 32-bit IEEE FP, 2 external buses |
|  | DSP96001 | 1990 | 60 or 75 | DSP96002 with 1 external bus |
|  | DSP56116 | 1991 | 50 | 16-bit version of DSP56001 |
|  | DSP56002 | 1993 | 50 | Enhanced DSP56001, with PLL on-chip |
| National Semiconductor | LM32900 | 1987 | 100 | 16/32-bit FX, no external memory |
| NEC | $\mu \mathrm{PD} 7720$ | 1980 | 250 | Heavily used, early FX DSP |
|  | $\mu$ PD7281 | 1984 | - | Dataflow DSP, for image processing |
|  | $\mu \mathrm{PD} 77230$ | 1985 | 150 | 32-bit FP |
|  | VISP | 1989 | - | For video, not marketed |
|  | $\mu$ PD6380 | 1990 | 122 | Update of 7720 , for digital audio |
| Oki <br> Semiconductor | MSM6992 | 1986 | 100 | 22-bit FP |
| SGS-Thompson | ST18930 | 1988 | 80 | 16-bit, integer or complex maths |
| Toshiba | 6386/7 | 1983 | 250 | 16/31-bit FX |
| Sharp | LH9124 | 1991 | 25 | 24-bit block FX, time/freq. domain |
|  | LH9320 | 1991 | - | Address generator for LH9124 |

Table A-1: Summary of important DSP devices from VLSI manufacturers. ${ }^{\text {a }}$

| Company | Part No. | Release Date | MAC (ns) | Comments |
| :---: | :---: | :---: | :---: | :---: |
| Texas <br> Instruments | TMS32010 | 1982 | 390 | Early popular DSP, 16-bit FX |
|  | TMS32020 | 1985 | 195 | Update of TMS32010 |
|  | TMS320C25 | 1987 | 100 | CMOS update of TMS32020 |
|  | TMS320C30 | 1988 | 50 or 60 | 32-bit FP |
|  | TMS320C50 | 1990 | 35 or 50 | 16-bit FX |
|  | TMS320C40 | 1991 | 40 or 50 | Enhanced 'C30, 6 comms. ports |
|  | TMS320C80 | 1995 | 40 | Integrates 4 64-bit <br> FP DSPs with a <br> 32-bit RISC CPU |
| Zoom | ZM6420 | 1988 | 350 | 20-bit FX, serialbus ASP, on-chip ADC and DACs |
| Zoran | 34161 | 1986 | 100 | 16-bit block FP, high-level DSP instructions |
|  | 34322 | 1988 | - | 32-bit block FP |
|  | 35325 | 1989 | - | 32-bit IEEE FP |

Table A-1: Summary of important DSP devices from VLSI manufacturers. ${ }^{\text {a }}$
a. In this table, the abbreviations FX and FP are used for fixed-point and floating-point, respectively.

## Appendix B

## Topology Expansion Metrics

The tables below detail the expansion increments for the thirteen M-P topologies considered in Chapter 5. Where more than one metric appears, $b$ and $w$ take precedence over $l$ and $d$.

| Topology | Number of Nodes | Increment | Factor <br> Increase |
| :---: | :---: | :---: | :---: |
| full interconnection | $n$ | 1 | $\frac{n+1}{n}$ |
| shared bus | $n$ | 1 | $\frac{n+1}{n}$ |
| simple ring | $n$ | 1 | $\frac{n+1}{n}$ |
| chordal ring | $n$ | 2 | $\frac{n+2}{n}$ |
| $b$-ary tree | $\frac{b^{l}-1}{b-1}$ | $\frac{(b+1)^{l}-b^{l}-n}{b}$ | $\frac{(b+1)^{l}-1}{n b}$ |
|  |  | $b^{l}$ | $b+\frac{1}{n}$ |
| $b$-ary x-tree | $\frac{b^{l}-1}{b-1}$ | $\frac{(b+1)^{l}-b^{l}-n}{b}$ | $\frac{(b+1)^{l}-1}{n b}$ |
|  |  | $b^{l}$ | $b+\frac{1}{n}$ |
| nearest-neighbour mesh | $w^{\text {d }}$ | $(w+1)^{d}-w^{d}$ | $\left(\frac{w+1}{w}\right)^{d}$ |
|  |  | $(w-1) w^{d}$ | $w$ |
| spanning-bus hypercube | $w^{\text {d }}$ | $(w+1)^{d}-w^{d}$ | $\left(\frac{w+1}{w}\right)^{d}$ |
|  |  | $(w-1) w^{d}$ | $w$ |

Table B-1: M-P topology node expansion metrics.

| Topology | Number <br> of Nodes | Increment | Factor <br> Increase |
| :--- | :---: | :---: | :---: |
| dual-bus hypercube | $w^{d}$ | $(w+1)^{d}-w^{d}$ | $\left(\frac{w+1}{w}\right)^{d}$ |
|  |  | $(w-1) w^{d}$ | $w$ |
| torus | $w^{d}$ | $(w+1)^{d}-w^{d}$ | $\left(\frac{w+1}{w}\right)^{d}$ |
|  |  | $(w-1) w^{d}$ | $w$ |
| $R$-ary $N$-cube | $N R^{N}$ | $N\left[(R+1)^{N}-R^{N}\right]$ | $\left(\frac{R+1}{R}\right)^{N}$ |
|  |  | $[N(R-1)+R] R^{N}$ | $\frac{R(N+1)}{N}$ |
| binary hypercube | $2^{d}$ | $2^{d}$ | 2 |
| cube-connected cycles | $d 2^{d}$ | $(d+2) 2^{d}$ | $\frac{2(d+1)}{d}$ |

Table B-1: M-P topology node expansion metrics.

| Topology | Number of Links | Increment | Factor Increase |
| :---: | :---: | :---: | :---: |
| full interconnection | $\frac{n(n-1)}{2}$ | $n$ | $\frac{n+1}{n-1}$ |
| shared bus | 1 | 0 | 1 |
| simple ring | $n$ | 1 | $\frac{n+1}{n}$ |
| chordal ring | $\left\lfloor\frac{3 n}{2}\right\rfloor$ | $(n \% 2)+1$ | $\frac{(n \% 2)+1}{\left[\frac{3 n}{2}\right]}+1$ |
| $b$-ary tree | $n-1$ | $\frac{(b+1)^{l}-b^{l}-n}{b}$ | $\frac{(b+1)^{l}-(b+1)}{b(n-1)}$ |
|  |  | $b^{l}$ | $\frac{b^{l}-1}{b^{I-1}-1}$ |

Table B-2: M-P topology link expansion metrics.

| Topology | Number of Links | Increment | Factor Increase |
| :---: | :---: | :---: | :---: |
| $b$-ary x-tree | $2(n-1)$ | $\frac{2\left[(b+1)^{l}-b^{l}-n\right]}{b}$ | $\frac{(b+1)^{l}-(b+1)}{b(n-1)}$ |
|  |  | $2 b^{l}$ | $\frac{b^{l}-1}{b^{l-1}-1}$ |
| nearest- <br> neighbour <br> mesh | $(w-1) d w^{d-1}$ | $w d\left[(w+1)^{d-1}-(w-1) w^{d-2}\right]$ | $\frac{(w+1)^{d-1}}{(w-1) w^{d-2}}$ |
|  |  | $(w-1) w^{d-1}[w(d+1)-d]$ | $\frac{w(d+1)}{d}$ |
| spanning- <br> bus <br> hypercube | $d w^{d-1}$ | $d\left[(w+1)^{d-1}-w^{d-1}\right]$ | $\left(\frac{w+1}{w}\right)^{d-1}$ |
|  |  | $w^{d-1}[w(d+1)-d]$ | $\frac{w(d+1)}{d}$ |
| dual-bus hypercube <br> torus | $2 w^{d-1}$ | $2\left[(w+1)^{d-1}-w^{d-1}\right]$ | $\left(\frac{w+1}{w}\right)^{d-1}$ |
|  |  | $2(w-1) w^{d-1}$ | $w$ |
|  | $d w^{d}$ | $d\left[(w+1)^{d}-w^{d}\right]$ | $\left(\frac{w+1}{w}\right)$ |
|  |  | $w^{d}[w(d+1)-d]$ | $\frac{w(d+1)}{d}$ |
| $R$-ary $N$-cube | $N R^{N+1}$ | $N\left[(R+1)^{N+1}-R^{N+1}\right]$ | $\left(\frac{R+1}{R}\right)^{N+1}$ |
|  |  | $R^{N+1}[R(N+1)-N]$ | $\frac{R(N+1)}{N}$ |
| binary hypercube | $d 2^{d-1}$ | $(d+2) 2^{d-1}$ | $\frac{2(d+1)}{d}$ |
| cube- <br> connected cycles | $3 d 2^{d-1}$ | $3(d+2) 2^{d-1}$ | $\frac{2(d+1)}{d}$ |

Table B-2: M-P topology link expansion metrics.

## Appendix C

## Topology Analysis Tool

In order to compare altemative M-P topologies, the author has developed a stand-alone add-in for Excel. Written in C, this .XLL implements the topology metrics discussed in Chapter 5.

## C. 1 Overview

This appendix lists the definition, include and source files required to re-build $T P G$ - the Topology Analysis Tool - which automatically registers the following worksheet functions:

| Topology | Function Name | Arguments $^{\text {a }}$ |
| :--- | :--- | :--- |
| full-interconnection | fulc | n, metric ${ }^{\text {b }}$ |
| shared bus | bus | n, metric |
| simple ring | ring | n, metric |
| chordal ring | c_ring | n, metric |
| $b$-ary tree | tree | b, l, metric, [expansion] ${ }^{\text {c }}$ |
| $b$-ary x-tree | x_tree | b, l, metric, [expansion] |
| nearest-neighbour mesh | mesh | w, d, metric, [expansion] |
| spanning-bus hypercube | sb_hpc | w, d, metric, [expansion] |
| dual-bus hypercube | db_hpc | w, d, metric, [expansion] |
| torus | torus | w, d, metric, [expansion] |
| $R$-ary $N$-cube | rn_cube | R, N, metric, [expansion] |
| binary hypercube | b_hpc | d, metric |
| cube-connected cycles | cc_cycles | d, metric |

Table C-1: Worksheet functions supported by $T P G$.

[^52]
## C. 2 Definition Files

The Topology Analysis Tool .XLL definition file is listed below.

## C.2.1 tpg.def



## = = = = = = = = = = Pre-amble $==========$

DESCRIPTION 'Add-ins functions for calculating M-P topology metrics' EXETYPE WINDOWS
CODE PRELOAD MOVEABLE DISCARDABLE DATA PRELOAD MOVEABLE SINGLE
$\begin{array}{ll}\text { HEAPSIZE } & 8192 \\ \text { STACKSIZE } & 0\end{array}$

## C. 3 Include Files

Topology Analysis Tool includes are listed in this section.

## C. 3.1 tpg_alg.h




ALG_SUCCESS

/* b, w or R
$/ * \mathrm{l}, \mathrm{d}$ or $\mathrm{N} * /$




 Finclude "tpg_utl.h"


 \#include <tetio.h>
\#include "tpg_utl.h"
typedef enum AlgErrCod_








C-7




Btatic
AlgErrCod
alg_f_frtlinkinc






C-11

C-12







## C.3.2 tpg_inf.h










## C. 3.3 tpg_ini.h


/* TPG includes
$\begin{array}{ll}\text { \#include } & \text { "tpg_alg.h" } \\ \text { \#include } & \text { "tpg_utl.h" }\end{array}$


/ C : enum for INI error codes
ypedef enum IniErrcod
typedef enum IniErrCod_
$\begin{array}{ll}\text { /* success } & \text { */ } \\ \text { /* Eatal error } & \text { */ } \\ \text { /* missing required XLOPER */ }\end{array}$

typedef onum xlotypCod_
$\mathrm{xt}_{-}$RQD $=0$.



IniExrCod
ini f_xlointgtz




## C. 3.4 tpg_utl.h






## C. 3.5 tpg_win.h





## C.3.6 tpg xlo.h



[^53]


\# define XLO_STR_MAX 256



typedef enum XloErrcod_
xLo_success
\} XloErrcod;




xloErrCod
xlo_f_eetxlostr


## C. 4 Source files

Source code for the Topology Analysis Tool is listed below.

## C.4.1 tpg_alg.c


$1 *==========$
$1 *$ Cheaders
$1 *========$
$\begin{array}{ll}\text { \#include } & \text { <math.h> } \\ \text { \#include } & \text { <stdlib.h> }\end{array}$

$\begin{array}{ll}\text { \#include } & \text { "tpg_alg.h" } \\ \text { \#include } & \text { "tpg_utl.h" }\end{array}$





metric value
outputs : pdMetval

AlgErrcod
alg f_fulintion
$\begin{array}{ll}\text { int } & \text { netricCod } \\ \text { nMetCod, }\end{array}$
MetricCod $\quad$ nMetcod,
double
\{ /* orror checking */
RETURNS : AlgErrCod
SELF
alg chkintgtz
..
alg_m_chknulptr! -1_-

alg_m_chkintgtz ( $n$ );
alg_m_chknulptr (pdMetVal);
/* choose topology metric */
/* choose topology metric */
switch (nMetcod)
case MT_NUM_NOD: /* number of nodes */
(*pdMetVal) $=n$; (paMatVal)
\} case MT_NUM_CON:
(* number of connections */
(*pdMetVal) $=n *(n-1.0) ;$ break; case MT_NUM_LNK:
/* number of links */
(*pdMetVal) $=n *(n-1.0) / 2.0$; break;
$\left\{\begin{array}{l}\text { /* node expanaion increment */ } \\ \text { (*pdMetVal) }=1.0 ;\end{array}\right.$ break:
(CABE MT_NOD_FAC:



Case MT_DV_PROD:

AlgErrCod
alg_f_sharedbus
$\begin{array}{lc}\text { int } & \text { n, } \\ \text { Metriccod } & \text { nMetCod, } \\ \text { double } & \text { *pdMetVal }\end{array}$
/* error checking */
alg_m_chkintgtz( n
alg_m_chknulptr ( pdMetVal); ;



alg_m_cllalgfnc( alg_f_chordlrng( $n, M T_{-} T O P_{1} D T A$, \&top_dia) $)$ alg_m_cllalgfnc( alg_f_chordlrng( $\left.n, ~ M T P_{-} V A L E N C Y, ~ \& v a l e n c y\right)$ ); (*pdmetVal) = top_dia * valency; \} break;


$$
\begin{aligned}
& \begin{array}{l}
\text { /* number of connections */ } \\
\text { (*pdMetVal) }=2.0 * \text { floor }(3.0 * n) / 2.0) \text {; }
\end{array} \\
& \text { break; } \\
& \begin{array}{l}
\text { case MT_NUM_LNK: } \\
\text { \{ }
\end{array} \\
& \begin{array}{l}
\text { /* number of links */ } \\
\text { (*pdMetval) }=\text { floor }(3.0 * n) / 2.0) \text {; }
\end{array} \\
& \text { break; } \\
& \text { case MT_NOD_INC: } \\
& \begin{array}{l}
\text { /* node expansion increment */ } \\
\text { (*pdMetVal) }=2.0 \text {; }
\end{array} \\
& \text { break; } \\
& \text { case MT_NOD_FAC: } \\
& \begin{array}{l}
\text { /* node expansion factor */ } \\
\text { (*pdMetVal) }=(\mathrm{n}+2.0) / \mathrm{n} \text {; }
\end{array} \\
& \text { break; } \\
& \text { case MT_LNK_INC: } \\
& \begin{array}{l}
\text { /* link expansion increment */ } \\
\text { (*pdMetVal) }=\left(\begin{array}{l}
\text { ( }
\end{array}\right)+1.0 \text {; }
\end{array} \\
& \text { break; } \\
& \begin{array}{l}
\text { /* link expansion factor */ } \\
\text { (*pdMetVal) }=1.0+((n+2)+1.0) / \text { floor }(3.0 * n) / 2.0) \text {; }
\end{array} \\
& \text { \} break; } \\
& \begin{array}{l}
\text { /* link conectivity */ } \\
\text { (*pdMatval) = } 3.0 ;
\end{array} \\
& \text { \} break; }
\end{aligned}
$$



$(*$ pdMetVal $)=2.0 *(n-1.0)$;

- node fanout

- number of levels
- topology metric code
- expansion parameter code
- metric value


AlgErrCod
alg_f_fulingtre
int
int
Metric
Expnen
double

alg_m_chkintgtz( $b$
alg_m_chkintgtz(
alg_m_chknulptr( pdMetVal); $; ~$
/* choose topology metric */
switch ( nMetCod)
case MT_NUM_NOD:
/* number of
Ol $^{180}$
$1 *$ dofault case */
$(*$ pdMetVal) $=($ pow $(b, 1)-1.0) /(b-1.0)$;
break;



/* number of connections */
double $n=0.0$;
alg_m_cllalgfnci alg_f_fulrngtrel $b$,
1,
MT_NUM_NOD
nExpCod,
(an) ); (*pdMetVal) $=4.0 *(n-1.0)$
break;
\}case MT_NUM_LNK:
$\{$ /* number of 1
 (*pdMetVal) $=2.0 *(n-1.0)$;

[^54]

[^55]

[^56]AlgErrCod
alg_f_duab
alg_m_cllalgfne( alg_f_dbhlnkinc( $w, d, \quad$ nexpcod, pdMetVal));
break;






double top dia $=0.0 ;$
double valency $=0.0 ;$


alg_m_cllalgfnc( alg_f_rncnodinc( $R, N$, nexpCod, pdmetVal) ):
 ¢abe MT_LNK_FAC:
/* link expansion factor */
alg_m_Cllalgfncl alg_f_rnclnkfac( R, N, nExpCod, pdMatVal) );




AlgErrcod
alg_f_cubco
(
AlgErrcod
alg_f_cubconeyc
(
$\begin{array}{lc}\text { int } & \text { d, } \\ \text { Metriccod } & \text { nMetcod, } \\ \text { double } & \text { *pdMetVal }\end{array}$
/* orror checking */
alg_m_chkintgtz( d );
alg_m_chknulptr( pdMotval);
/* choose topology metric */
awitch ( nMetcod)
\{ case MT_NUM_NOD:
(*) number of nodes */ , breaki
(* number of connections */
(*pdMetVal) $=3.0 * d *$ pow $(2.0, d)$; (*pdMetVal) $=3.0 * d$ * pow ( 2
\} break;
cabe MT_NOD_INC:
\{ $/ *$ node oxpaneion increment */
(* number of linke */
(*pdMetval) $=3.0$ * $d$ pow (2.0, $d-1.0)$;






switch (nExpCod)



 return ( ALG_FTL_ERR) ;
/* incrementing fanout, $b$ */
(*pdMetVal) $=($ pow $(b+1.0,1)-\operatorname{pow}(b, 1)-n) / b ;$
break; $\quad$ ( pow (b, 1) -n n ) b ;

( /* incrementing levels, 1 */ (*pdMetval) $=\operatorname{pow}(\mathrm{b}, 1$ ); break;
Case EX_FRST:
tatic
return( AlG_sUCCESS) ;

[^57] return ( ALG_FTL_ERR);

\}
return

tatic
alg_f_nnminkinc
Expnen
alg m chkintgtz (w )
alg_m_chkintgtz ( d )
/* choose expansion parameter */
switch (nExpCod)
case EX_FRST






static
AlgErrcod

$\underset{\text { alg_m_chkintgtz }}{\text { /* }}$ w

alg_m_chkintgtz( d $) ;$
alg_m_chknulptr( pdMetval) ;

return ( ala_success);










## C.4.2 tpg_inf.c



| /* SDK headers |  |  |  |  |  |
| :---: | :---: | :---: | :---: | :---: | :---: |
|  |  |  |  |  |  |

 \#include
include
incluall.h>
<framewrk,
\#include <framewrk.h>



\{" inf_x_fulintcon",
/* choose expansion parameter */
switch (nExpCod)

return ( ALG_FTL_ERR);

[expansion]",
[expansion]",
[expansion]",
[expansion]",





/* return XLOPER */
return( \&xResult);
d,
metric,
expansion expansion, /* free all XLOPERs passed in by Excel */
/* free all XLOPERs passed in by Excel Excel (xlFree, 0,1, epxDimnsn)
 /* return XLOPER */
return( \&xResult);


## LPXLOPER far pascal <br>  <br> LPXLOPER pxWidth, LPXLOPER pxDimnen <br> LPXLOPER pXMetric, <br> xResult ; Metriciod static XLOPER int int <br>  <br> $\qquad$ <br> LPXLOPER <br> $\qquad$


double dMetval $=0.0$;
/* clear return XLOPER */
/* clear return XLOPER */
memset( \&xResult, 0, sizeof( xResult));
/* test input XLOPERs and init standard $v$
1* test input XLOPERa and init standard
ini m_xlointgtz ( pxDimnen, d $\quad$,
ini_m_xlometcod( pxMetric, metric
/* call topology analysis algorithm */
/* call topology analysis algorithm */
tpg_m_cllalgfnc ( alg_f_spabushpc (w, d,
/* set return XLOPER */
xlo_m_setxlonum ( dMetval, \&xResult);
(* free all XLOPERs passed in by Excel */
Excel ( xlFree, 0, 1, \&pxWidth );
Excel ( xlFree, 0, 1, \&pxDimnen) ;
 Excel ( xlFree, 0, 1, \&pxExpnen) /* return XLOPER */
return ( \&xResult);


LPXLOPER far pascal
inf $\times$ duabuahpe
LPXLOPER pxWidth,
pxMetric
$\begin{array}{ll}\text { LPXLOPER } & \text { pXMetric } \\ \text { LPXLOPER } & \text { pxExpron }\end{array}$





## C. 4.3 tpg_ini.c



LPXLOPER pxResult
static
XLOPER
XLOPER xInteger;
/* clear atatic XLOPER */
memset ( \&xInteger, 0, sizeo
/* check XLOPER is not missin /* check XLOPER is not missing */
ini_m_chkxlomis( pxInput, nArgTyp, szInput, pxResult); /* coerce XLOPER to an integer
Excel (xlCoerce /* coerce XLOPER to an integer */ Excel ( xlCoerce, \&xInteger, 2, pxInput, TempInt( xltypeInt)); /* check XLOPER is not an Excel \#error */
ini_m_cllinifnc (ini_f_chkxloerr( \&xIntege /* get integer value from coerced XLOPER */ (*pinteger) = xlo_m_getxloint ( \&xinteger); ini_m_cllinifnc( ini_f_chkxloerr( \&xInteger, szinput, pxResult)); /* get integer value from coerced XLOPER */ /* convert integer value */ ini_m_ellinifnc( ini_f_chkintgtz( (*pInteger), szInput, pxResult)) return ( INI_sUCCESS);
ini_m_cllinifnc( ini_f_cnvmetcod saMetCod, azInput, pMetCod, pxResult)); return( INI_sUCCESS);

IniErrcod
ini_f_xloexpeod
const LPXLOPER





- metric code initialised from XLOPER


Expnencod *pexpcod,
LPXLOPER pxResult


[^58]

const char *szInput,




## C. 4.4 tpg_utl.c



\#include "tpg_utl.h"

UtlErrCod
utl_f_cree
int nExpIdx = EX_FRST;
/* check for NULL ptrs */
ini_m_chknulptr( BzString)
ini_m_chknulptr( ezInput);
ini_m_chknulptr( pnExpCod);
/* compare expansion increment with known strings */
for ( nExpIdx = EX_FRST; nExpIdx <=EX_SCND; nExpIdx++) if (ini_f_matstrcod ( sestring,
 (*pnExpCod) = nExpIdx;
return ( INI_success);
utl_f_wrierrlogi $n * *$ unknown expansion parameter: $\boldsymbol{t s}_{8}=t 8 \backslash \mathbf{n}^{\prime \prime}$,
azInput;
szString) ;

xlo_f_setxlostr( pxResult,
xlo_f_setxlostr( $\left.\underset{\text { "<tes> }}{\operatorname{pxResult}}(=\}_{B}\right)$ is an unknown expansion parameter" szInput,
sestring);
return ( INI_FTL_ERR);


*s2LogFil



## C. 4.5 tpg_win.c





\#include "tpg_inf.h"
\#include
\#include
\#include
\#include
\#include
*

* External definitions
/* External definitions
extern
const
LPSTR
rgStR




static XLOPER BLatic
XLOPER

$$
\begin{array}{lll}
\text { int } & \text { i } & =0 ; \\
\text { char } & \text { BzName [XLO_STR_MAX] } & =" " ;
\end{array}
$$

xlo_m_eetxloerr ( xlerrValue, \&xRegId);

$$
\begin{aligned}
& \text { /* re-register functions in add-in */ } \\
& \text { for }(i=0 ; i<\text { INF_FNC_RWS; i++ })
\end{aligned}
$$

$$
\begin{aligned}
& \text { for }(i=0 ; i<\text { INF_FNC_RWS; } i++) \\
& \text { \{lo_f getxlobtr }(\text { pxName, gzName ); }
\end{aligned}
$$

$$
\begin{aligned}
& \text { if ( !atremp ( rgFunce }[i][0], \text { szName }) \text { ) } \\
& \text { Excel ( xlGetName, \&xDLL, } 0) \text {; }
\end{aligned}
$$

$$
\begin{gathered}
\text { Excel( xlfRegister, } \\
\text { \&xRegId, }
\end{gathered}
$$

INF_FNC_CLS +1 ,

Tempstri( rgFunce
TempStri rgFunce
TempStr rgFunce
TempStry rgFunce [i] (2]),
TempStr( rgFunce[i][3]),
TempStr( rgFunce[i] [4]),

Tempstr( rgFunca[i](6%5D)),
Excel (xlFree, 0, 1, \&xDLL); return ( \&xRegId);

NOTES : Add XLL to start-up. INI file using Excel add-in manager INPUTS INPUTS : <none>
return( \&xRegId);

RETURNS : int far pascal

## C.4.6 tpg_xlo.c




 \#include $\quad$ "tpg_xlo.h"
\#include

[^59]



## Appendix D

## Digital Console Scheduler

In order to compare the performance of scheduling strategies to DMCs, the author has developed a scheduler specifically targetted at the DMC problem domain. This application is coded in the strongly-typed compiled Prolog implementation available from Borland.

## D. 1 Overview

This appendix lists the definition and source files required to re-build DCS - the Digital Console Scheduler. Table D-1, below, illustrates the various options supported by the DCS scheduling engine for static-level (CPM), dynamic-level (DYN) and A* algorithms (AST).

| Algorithm ${ }^{\text {a }}$ | IPC Mechanism | Static Labelling | Dynamic Labelling | Heuristic Underestimate |
| :---: | :---: | :---: | :---: | :---: |
| CPM | $\begin{aligned} & \text { FWE (d) } \\ & \text { NSL } \\ & \text { NML } \end{aligned}$ | HLN <br> HLE (d) <br> LCN <br> LCE <br> RND |  |  |
| DYN | $\begin{aligned} & \text { FWE (d) } \\ & \text { NSL } \\ & \text { NML } \end{aligned}$ | HLN HLE (d) LCN LCE RND | NON <br> PRO <br> TSK (d) <br> ALL |  |
| AST | $\begin{aligned} & \text { FWE (d) } \\ & \text { NSL } \\ & \text { NML } \end{aligned}$ |  |  | NON <br> MPC <br> MIA <br> MRC <br> ALL (d) |

Table D-1: Options supported by the $D C S$ scheduling engine.
a. (d) indicates that this is the default value for the given option.
The Digital Console Scheduler public definition files are listed below.


## D.2.2 des_clc.pub








| /* | : ThroughPut |
| :---: | :---: |
| /* |  |
| /* CALLS | : clc_fin_tim(...) |
| /* | : idb_pro_exe (...) |
| /* | : lst_num_elm(...) |
| /* |  |
| /* database | -- |
| /* |  |
| /* BACKTRACKING : determ |  |
|  |  |
| determ cle_thr_put 1 |  |
|  |  |
|  |  |
| $\begin{aligned} & \text { d_tsk_lst, } \\ & \text { d_pro_lst, } \end{aligned}$ |  |
| d_thr_put |  |
| ) - ( $\bar{i}, i, 0)$ |  |
| /* |  |
| /* PREDICATE | : clc_tsk_grn (...) |
| /* |  |
| /* DESCRIPTION : Calculate granularity statistics for <Tasklist> |  |
| /** PARAMETERS | TaskList |
| /** | : TaskList |
| /* | : MaxGran |
| /* | : AvgGran |
| /* |  |
| /* CALLS | : clc_ctg_aux(...) |
| /* | : clc_seq_sch (...) |
| /* | : idb tsk_exe (...) |
| /* | : lst_num_elm(...) |
| /* ${ }^{\text {/* }}$ ( ${ }^{\text {a }}$ |  |
| /* DATABASE | : -- |
|  |  |
| /* backtracking : determ |  |
| /* ---------- |  |
|  |  |
| clc_tsk_grn |  |
|  |  |
| d_tsk_lst, |  |
| d_tsk_exe, |  |
| d_tsk-oxe, |  |
| d_tkf_grn |  |
| ) - (ī, 0, o, o) |  |
| /* --. |  |
| /* Predicate | : clc_trk_par (...) |
| /* |  |
| /* Description | : Calculate critical-path, <CritPath>, and taskforce |
| /* | : parallelism, <TaskPara>, for <TaskList> with <Proclist> |
| /** | : precedence relations |
| /* |  |
| /* Parameters | : Tasklist |

## D.2.3 des_cpm.pub



## D.2.4 des_dyn.pub





[^60]| /* CALLS | : edb_ebd_aux (...) |
| :---: | :---: |
| /* |  |
| 1* Database |  |
| /* |  |
| /* BACKTRACKING : determ |  |
|  |  |
| ```determ edb_btr_dsp 1``` |  |
|  |  |
|  |  |
| $\begin{aligned} & \text { db-selector, } \\ & \text { file } \end{aligned}$ |  |
| ) - (i, i) |  |
| /* |  |
| /* predicate | : edb_dba_ste ( . . ) |
| /* |  |
| /* DESCRIPTION : Write statistics from external DB <DbaSel> to <OutDev> |  |
| /* PARAMETERS | DbaSel |
| /* | OutDev |
| /* |  |
| /* calls | -- |
| /* |  |
| /* DAtabase | db_statistics(...) |
| /* |  |
| /* baCKtracking : determ |  |
| /* ------------------- |  |
| determ edb_dba_日tв 1 |  |
| $\begin{aligned} & \mathrm{db} \text { selector, } \\ & \text { file } \end{aligned}$ |  |
| ) - (i, i) |  |
| /* ------.------------------ |  |
| /* PREDICATE | edb_btr_sta (...) |
| /* - - |  |
| /* ${ }^{\text {/ }}$ | Write statistics of all btrees in external DB <DbaSel> to |
| /* |  |
| /* Parameters | Dbasel |
| 1* | OutDev |
| /** |  |
| /* calls | edb_ebs_aux (...) |
|  |  |
| /** Database : ab_beree日(...) |  |
| /* BACKTRACKING : determ |  |
| 1* --- |  |
| determ |  |
| edb_btr_bte |  |
| db_selector, |  |

## D.2.5 dcs edb.pub



## D.2.6 des_glb.pub


$========$
constants
$=========$

file
$(i, i)$


* number of instruction cycles for IPC
c_ipc_tim $=2$
* character codes for ESCAPE and RETURN keys $c_{-k e y}$ esc $=27$
$c_{-}$key_rtn $=13$
8nulll string and path delimiter
c_nul_str $=$ n"
c_pth_dlm $=" \backslash \backslash "$
system-wide sampling rate
c_smp_rat $=48000$
zero offset for dynamic level btree keys
c_off_set $=100000$


* directory, filename, extension and symbolic filenames
d_fil_nam $=$ etring $\begin{aligned} \text { d_dos_dir } & =\text { string } \\ \text { d_dos_ext } & =\text { string }\end{aligned}$
* scheduling statistics
 $\begin{aligned} & \text { d_sch_spd }^{\text {d_ser }}=\text { real } \\ & \text { d_sys_tim }^{\text {a }}\end{aligned}$
$\begin{aligned} \text { d_eys_tim }^{\text {dithr }} & =\text { string } \\ \text { d_thr_put } & =\text { real }\end{aligned}$
$\begin{aligned} d_{\text {_tkf_grn }} & =r \text { real } \\ \text { d_tkf_par } & =r e a l\end{aligned}$
t task levels, critical path, list counte and return status d cpm lvl $\begin{aligned} \text { d_crt_pth } & =\text { integer } \\ \text { d_dyn_lvi } & =\text { integer }\end{aligned}$
d_num_elm $=$ real
$\begin{aligned} d_{\text {_num_ncx }} & =\text { real } \\ d_{-} \text {rtn_stt } & =\text { integer }\end{aligned}$

D-11

## D.2.7 des_idb.pub



* hardware
proc ( d_pro_nam, d_pro_typ)
proctype( d_pro_typ, d_pro_exe)
* general algorithm facte

* sequential cost and algorithm run-time
dbatime( integer, integer, integer, integer)
** internal facts
node( d_tak_lst, d_pro_lst, d_exe_tim)
num_sea_nod ( real)

/* $\begin{aligned} & \text { global predicater } \\ & ==================\end{aligned}$



D.2.8 des_Ist.pub

/* ***********************


[^61]/* BACKTRACKING : determ

determ
lat_fat_elm
d_tak_lst
leterm







determ
determ
lat mem

determ
1st_mem_elm
( diprerel.
d_pre_rel,
d_pre_let
$(\bar{i}, i)$,
$\underset{\substack{\text { determ } \\ \text { lat_mem_elm }}}{\substack{\text { din }}}$






[^62]
nondeterm

) - (i, o)








## D.2.9 des_sch.pub




## D.2.10 des_trl.pub


$====3=3====1$
global predicatos
$===============$



| /* predicate | tri_ipc_sym(...) | */ |
| :---: | :---: | :---: |
| /** |  | * |
|  | Translate <IpcCod> to IPC scheme string <AlgStr> | */ |
| /* parameters | Ipccod | */ |
| /* | Ipcstr | */ |
| /* calls |  | */ |


determ
sch_cmp_idl
d_pro_lst,
d_fin_tim,
d_pro_lst
$(\bar{i}, i, ~ o)$




## D.2.11 des_utl.pub






determ
utl_lbl






## D.2.12 dcs_wri.pub






[^63]

## D. 3 Private Definition Files

## $D C S$ private predicate definition files are listed in this section.

## D.3.1 des_ast.prv



determ
d_tak_lat,
d_pro_lat


BACKTRACKING : non-determ

| /* DESCRIPTION <br> /* | : Generate a successor to <OldNode>, using <IPCScheme> |
| :---: | :---: |
| /* Parameters | : Ipescheme |
| /* | : OldNode |
| /* | : Node |
| /* |  |
| /* CALLS | : idb_tak_exe (...) |
| /* | : lat_del_ndt (...) |
| /* | : BCh_act_tak (...) |
| /* | : ach_ins_idl (...) |
| /* | : Bch_ins_ipe(...) |
| /* |  |
| /* | : BCh_non_cor (...) |
| /* | : sch_pre_rel(...) |
| /* | : sch_pro_rsce(...) |
| /* |  |
| /* Database | : -- |
| /* |  |
| /* BACKTRACKING : non-determ |  |
| /* --------- |  |
| nondeterm <br> ast_suc_nod |  |
| d_ipc_cod, <br> d_ast_nod, <br> d_ast_nod, <br> d_nod_cet |  |
| $)$ ) |  |
| /* |  |
| /* Predicate | : ast_ins_new(...) |
|  |  |
| /* DESCRTPTION | : Insert <Node> after calculating heuribtic underestimate |
| /* | : given <IpcScheme> and <HrtScheme> and <Cost> of <Node> |
| /** |  |
| /* Parameters | : IpcScheme |
| /* | : HrtScheme |
| /* | : Btrsol |
| /* | : $\mathrm{GO}_{0}$ |
| /* | : Node |
| /* | : Cost |
| /* |  |
| /* Calls | : ast_hrt_val(...) |
| /* | : ast_ket_int (...) |
| /* | : ast_wat_dog (...) |
| /* - - |  |
| /* DAtabase | -- |
| /* |  |
| /* BACKTRACKING : determ |  |
| /* --. | -------------------------------------------------- |
| determ <br> ast_ins_new |  |
| d_ipe_cod, |  |




/* вACKTRACKING : determ







## D.3.3 des_cpm.prv





determ
epm_cre_sch
(
d_ipc_cod,
d_tsk_lst,
d_pro_lat,
d_fin_tim,
d_pro_lst

D.3.4 des_dyn.prv


/* /* backtracking :














| /* PREDICATE | : edb_ebd_aux (...) |
| :---: | :---: |
| /* |  |
| /* Description | : Auxiliary predicate for edb_btr_dep (...) |
| /* PARAMETERS | : DbaSel |
| /* |  |
| /* CALLS | : edb_wri_btre(...) |
| /* |  |
| /* Database | : db_btrees (. . . ) |
| /* |  |
| /* BACKTRACKING | : determ |
| /* ------ |  |
| determ |  |
| edb_ebd_.-2ux |  |
| db selector |  |
| $)$ - |  |
| 1* ------ |  |
| /* predicate | : edb_ebs_aux ( . . ) |
| /* |  |
| /* DESCRIPTION | : Auxiliary predicate for edb_btr_stb(...) |
| /* | - _ |
| /* parameters | : Dbasel |
| /* | : BtrNam |
| /* | OutDev |
| /* |  |
| /* CALLS | -- |
| /* |  |
| /* Database | : bt_open(...) |
| /* | : bt-statistics(...) |
| /* | : bt_close (...) |
| /* |  |
| /* BACKTRACKing | : determ |
| /* |  |
| determ |  |
| edb_ebs_aux |  |
| db_selector, d_btr_nam, file |  |
| ) |  |
| /* ------ |  |
| /* Predicate | edb_wri_btr(...) |
| /* |  |
| /* DESCRIPTION | : Write contents of all Btrees in the external DB <DbaSel> |
| /* PARAMETERS | : DbaSel |
| /* |  |
| /* CALLS | : edb_all_key(...) |
| /* Database | : db_btrees (...) |
| /* |  |






[^64]
## D.3.6 des inf.pry





| /** | : dyn_run_alg (...) |
| :---: | :---: |
| /* | : idb_sya_tim(...) |
| /* |  |
| /* database | -- |
| /* |  |
| /* BACKTRACKING | : determ |
| /* ---------- |  |
| determ inf_bgn_sch 1 |  |
|  |  |
|  |  |
| d_alg_ctl, <br> d_tsk_lst, <br> d_pre_lst, <br> d_pro_lst, <br> d_pro_lat |  |
| ) |  |
| /* --- |  |
| /* PREDICATE | : inf_get_inp(...) |
| /* |  |
| /* DESCRIPTION | : Get taskforce file <TkfFil>, hardware file <HdwFil> and |
| /* | : algorithm control parameters <AlgCtl> from usex |
| /* |  |
| /* PARAMETERS | : TkfFil |
| /** | : HdwFil |
| /* |  |
|  |  |
| /* CALlus | : inf_nul_inp(...) |
| /** | : inf_get_tkf (...) |
| /* |  |
| /* database | -- |
| /** BACKTRACKINa |  |
| /* BACKTRACKING : determ |  |
| determ |  |
| inf_get_inp |  |
| d_fil_nam, d_fil_nam, d_alg_ctl |  |
| ) - |  |
| /* -------. |  |
| /* Predicate | : inf_get_tkf(...) |
| /* |  |
| /** DESCRIPTION | : Using directory <Dir> and filename extension <Ext>, get |
| /** | : taskforce filname <Tkf> and then satisfy inf_get_hdw (...) |
| /* PARAMETERS | Dir - |
| /* | : Ext |
| /* | : OldInp |
| /* | : Inp |
| /** | : Ctil |
| /* |  |







| /** BACKTRACKING : determ |  |
| :---: | :---: |
|  |  |
| determ |  |
| inf_tat_tet |  |
| d_sym_lst |  |
| ) - |  |
| /* |  |
| /** PREDICATE : inf_ctp_tst(...) |  |
| /* DESCRIPTION : Bind <PassFail> to c_tet_fai if critical path will not fit |  |
| /** : on any processor, or c_tst_pas otherwise |  |
| /* parameters : Preclist |  |
| /* | : Proclist |
| /** : PassFail |  |
|  |  |
| /* calls | : cle_crt_pth(...) |
| /** : inf_det_aux (...) |  |
| 1* database |  |
| /* backiracking : doterm |  |
|  |  |
| determ |  |
| ${ }_{1}{ }^{\text {inf_ctp_tst }}$ |  |
|  |  |
| $\begin{aligned} & \text { d_pre_1st, } \\ & \text { d_pro_lst, } \\ & \text { d_tst_cod } \end{aligned}$ |  |
|  |  |
| /* ---.--- |  |
| /* predicate | : inf_det_aux ( . . ) |
| /* - - |  |
| /** DESCRIPTION : Auxiliary predicate for inf_ctp_tat (...) |  |
| /* phopmeter p procliot |  |
| /** |  |
| /* calls |  |
| /* ${ }_{\text {* }}^{\text {- }}$ 1db_pro_exe(...) |  |
|  |  |
| /** DATabase : - |  |
| /* backtracking : determ |  |
|  |  |
| determ |  |
| inf_det_aux |  |
| d_pro_lst, |  |

char,
d_dyn_cod






| /* /* BACKTRACKING: determ |  |
| :---: | :---: |
|  |  |
| determ |  |
| inf_sav_sch |  |
|  |  |
| , $\mathrm{d}_{-} \mathrm{rtn} \mathrm{s}_{-} \mathrm{tt}$ |  |
| - - |  |
| /** PREDICATE : inf_get_fill...) |  |
|  |  |
| /* DESCRIPTION : Get filename in which to store current dis output |  |
| /* PARAMETERS : |  |
|  |  |
| /* CALLS ${ }_{\text {/* }}$ : inf_vld_fil(...) |  |
| /** Database : -- |  |
|  |  |
| /* backtracking : determ |  |
|  |  |
| determ |  |
| inf_get_fil |  |
|  |  |
| 1 |  |
| /* |  |
| /* PREDICATE : inf_vld_fill...) |  |
|  |  |
| /* DESCRIPTION : Ensure output filename <SChFil> does not aleady exist |  |
| /* parameters : Schfil |  |
| /* |  |
| /* CALLS : inf_ren_fil(...) |  |
| /* database : -- |  |
| /* |  |
| 1* backtracking : determ |  |
|  |  |
| determ <br> inf_vld_fil |  |
| , d_fil_nam |  |
|  |  |
| /* -----.... |  |
| /* PREDICATE | : inf_ren_fill...) |
| /* description | : Roname temporary DCS output 'temp.och' to <SChFil> |
| /* | den |
| /* parameters | : Usichr |
| /* | : SchFil |

## D.3.7 des_Ist.prv



## D.3.8 des_sch.prv


determ
$\mathrm{sch}^{\mathrm{Ch}} \mathrm{PC}_{-}$non
d_fin_tim,
d_fin_tim,
d_tak_exe,



D.3.9 des_utl.prv



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## D.3.10 des_wri.pry


D. 4 Prolog Source Files
Source code for the Digital Console
Section D.4.7 lists the interface predicat
Source code for the Digital Console Scheduler is listed below.
Section D.4.7 lists the interface predicates including the main goal.

## D.4.1 des_ast.pro

/* **************************************************************************/,
/* 1995 : School of Engineering, University of Durham, Durham, England
/* Solid State Logic Ltd, Begbroke, Oxford, England
/*************************************************************************/,

$*$ VERSION: 8.7
$* * * * * * * * * * * * * * * * *)$

project "des.prj"
 include "des_glb.pub"

*



[^65]




| /* DESCRTPTION : Bind <MrCHrt> to MRC underestimate for <ProcList> *//*/* PARAMETERS |  |
| :---: | :---: |
|  |  |
| /* Parameters | : Proclist |
| /* | : Fintime |
| 1* | Oldmrchrt |
| /* | Mrchrt |
| /* |  |
| /* CALLLS | : SELF |
| /* | : ast mrc_rct(...) |
| /* | : clc_max_val (...) |
| /* | : idb_tsk_exe (...) |
| 1* | : idb_tsk_lvl (...) |
| /* - - |  |
| /* Database | -- |
| /* |  |
|  |  |
| /* BACKTRACKING : determ |  |
| ast_mrc_prol [], _, Mrchrt, MreHrt):- |  |
| ```ast_mrc_pro( [pr(_,TaskList)\|RestProce], FinTime, oldMrcHrt, MreHrt):- ast_met_rct( TaskList, Task, TaskFinTime), !, idb_tsk_lvl( Task, Level), idb_tsk_exe( Task, ExecTime), RemäningCP = Level - Exectime, Heuristic = (TaskFinTime - FinTime) + RemainingCP, clc_max_val( OldmrcHrt, Heuristic, NewMreHxt), ast_mxc_pro( RestProcs, FinTime, NewMreHrt, MrcHrt).``` |  |
| ```ast_mrc_pro( [_\|RestProce], FinTime, OldMrcHrt, MrcHrt):- ast_mrc_pro( RestProcs, FinTime, oldMreHrt, MreHrt).``` |  |
|  |  |
|  |  |
| /* |  |
| ```/* DESCRIPTION : Bind <FinTime> and <Task> to most recent task scheduled in /* : in <TaskList>``` |  |
|  |  |
|  |  |
| /* PARAMETERS : Tasklist |  |
| /* : Task |  |
| /** : Fintime |  |
| 1* |  |
| /* Calls : self |  |
| /* |  |
| /* Database : - - |  |
|  |  |
| /** BACKTRACKing : determ |  |
| /* ---------------- |  |
| ast_mst_rct ( [tk(idle,_)], _, _):- !, fail. |  |
|  | $k \text { (Task, FinTime) \|_], Task, FinTime):- }$ 10", TaBk <> "ipc", !. |
| ast_mat_rct ( [t Task <s "id | (idlo, Fintime), tk(Task, _) \|_], Task, FinTime):e", Task <> "ipc". |





## D.4.2 des_clc.pro







* database
 include "dcs_glb.pub"
include "dce_clc.pub"
include "dcs_cpm.pub"
include "dcs_idb.pub"
include "dcs_lst.pub"
include "dcs_sch.pub"
include "dcs_utl.pub"
include "dcs_wri.pub"
 include "dce_glb.pub"




[^66]$\begin{aligned} \text { DESCRIPTION } & : \text { Begin static labelled <LblScheme> algorithm to echedule } \\ & \text { : <TaskList> taska with <PrecList> precodence on <ProcList }\end{aligned}$

## D.4.3 des_cpm.pro





## D.4.4 des_dyn.pro







str_int( Key, olddynlvi),


## D.4.5 dcs_edb.pro


project "dcs.prj"

include "dcs_glb.pub"
include "dcs_edb.pub"


include "des_edb.prv"

database


[^67]


/* BACKTRACKING : determ


## D.4.6 des_idb.pro





 idb_tsk_exe ( TaskName, ExecTime):-
retract ( task (TaskName, TaskType)), !,
assertz (task (TaskName, TaskType)),
retract (tasktypei( TaskType, ExecTime, Inputs, outputs)), !,
assertz( tasktype( TaskType, ExecTime, Inputs, Outputs)).

キ* *

[^68]_tsk_lvi ( TaskName, Level):-
retract ( level ( TaskName, Level)). !,
assertz ( level ( TaskName, Level)).
PREDICATE
/**

## D.4.7 des_inf.pro



## project "des.prj"

 include "dce_gib.pub"
include "dce_glb.pub"
include "des_cle.pub"

| 0 |
| :---: |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |
| 0 |

include "ddes_dyn.pub"
include "dce_ idb.pub"
include "dcs 1 - $\mathrm{st} \cdot \mathrm{pub}$ "
include "dcs_tri.pub"
include "dce_utl.pub"

retract ( proc ( ProcName, Proctype)), retract ( proctype( ProcType, ProcRarc)),





[^69]



inf_mnu_chk( s_mnu_alg, UsrChr, ItmCod):-
inf_vld_alg( UsrChr, ItmCod), !.
 inf_mnu_chk( s_mnu_lbl, UsrChr, ItmCod):-
inf_vld_lbl( Usichr, ItmCod) inf_mnu_chk( s_mnu_dyn, UsrChr, ItmCod):inf_vld_dyn( Uarchr, ItmCod), !. inf_mnu_chk( s_mnu_hrt, Usechr, ItmCod):-

 inf_get alg( act_, Tpe, Lbl, Dyn, Hrt), Ctl, Dir, Ext, Oldinp, Inp):-








 inf_get_alg ( _, Ctl, Dir, Ext, ai(Tkf, Hdw), Inp) :clearwindow(), ${ }^{\text {c/win_two, } c_{-} \text {win_ref) }}$




/* BACKTRACKING : determ




inf_ctp_tst( PrecList, ProcList, s_tet_pas)


[^70]


[^71]
## D.4.8 des_Ist.pro


project "des.prj"

 include "des_glb.pub"
include "des_dym.pub"
include "des_lst.pub"


| include "dce_let.pry" |  |
| :---: | :---: |
| /* == = = = = |  |
|  |  |
| /* |  |
| /* PREDICATE | : lat_fat_elm(...) |
| /* |  |
| /* DESCRIPTION | : Extract first <Elom> from <List> |
| /* parameters | : Elem |
| /* | : List |
| /* |  |
| /* Calls | : -- |
| /* |  |




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Ist_apd_ndt(List2, WorkList, List1),
WorkList $=$ [Head/Tail2],
Worklist $=$ (Head/Tail2],
lst_apd_ndt( List2, Tail2, List3),
lst_per_mut( List3, Taili).

D.4.9 dcs_sch.pro


## project "des.prj"



include "dcs_cle.pub"
include "dcs idb.pub"
include "dcs_idb.pub"
include "dca_sch.pub"
 include "dcs_sch.prv"







| /* CALLS $/ *$ sch_ins_tsk(...) |  |
| :---: | :---: |
|  |  |
| /* DAtabase : -- |  |
| /* backtracking : determ |  |
|  |  |
| /* ----------------------- |  |
| ```sch_ipc_src( IpcStt, IpCEnd, pr(P,[tk(idle,TF)\|Done]), oldPL, PL, OldF, F):- IpeStt > TF, !, TaskList = [tk(ipe,IpoEnd), tk(idle,IpcStt)|Done], sch_ins_tsk( pr(P, TaskList), OldPL, PL, OldF, F).``` |  |
| ```sch_ipc_src( IpcStt, IpcEnd, pr(P,[tk(T,TF)\|Donel), oldPL, PL, OldF, F):- IpeStt > TF, !, TaskList = {tk(ipc,IpcEnd), tk(idle,IpeStt), tk(T,TF)|Done], sch_ins_tsk( pr(P, TaskList), Olapl, PL, OldF, F).``` |  |
| sch_ipc_src (_, IpcEnd, pr(P,Done), OldPL, PL, OldF, F):- <br> sch_ins_tak( prip, [tk(ipc, Ipcend)\|Done]), OldPL, PL, OldF, F). |  |
| /* |  |
| /* PREDICATE : sch_ipc_dst(...) |  |
|  |  |
| /* DESCRIPTION : Schedule IPC from <IpeStt> to <IpCEnd> on the destination |  |
| /* | : processor, binding <TaskList> to the result |
| /* Parameters : Ipcstt |  |
| /* | Ipcend |
| /* | OldTasklist |
| /* | TaskList |
| /* : TaskList |  |
| /* calls |  |
| /* |  |
| /* Database |  |
| /* |  |
| /* BACKTRACKING : determ |  |
| /* |  |
| ```sch_ipc_dst( IpeStt, IpcEnd, [tk(idle,TF)\|Done], TaskList):- IpcStt > TF, !, TaskList = [tk(ipc,IpcEnd), tk(idle,IpeStt)|Done].``` |  |
| sch_ipc_dst( IpcStt, IpcEnd, [tk(T,TF)\|Done], TaskList):IpeSte > TF, : <br> TaskList $=$ [tk(ipc,IpcEnd), tk(idle,IpcStt), tk(T,TF) |  |
| sch_ipc_dst ( _. Ipcend, OldTL, [tk(ipc, Ipcend)\|OldTL]). |  |
| / |  |
| /* PREDICATE : sch_ins_tsk(...) |  |
|  |  |
| /* DESCRIPTION : Insert a task on |  |
| /* | at <oldFinTime> to |
| /************) | the finish time < |
| /* Parameters | Proc |

IpcStt $=$ oldIpcstt,
IpcEnd $=$ OldIpcEnd.


BCh_ipc_chk( IpcStt, IpcEnd, [_|RestProcs])
Bch_ipc_chk( IpeStt, IpcEnd, RestProcs)


D.4.10 des_trl.pro

project "dce.prj"

/* Public predicate definitions
/* ====== $===================n=$
include "dce_glb.pub"
include "dce_trl.pub"

trl_alg_日ym( B_alg_cpm, "CPM: static priority list"):



| /* calls | : SELF | */ |
| :---: | :---: | :---: |
| /* | : lst_apd_lst (...) | */ |
| /* | : utl spl tkl (...) | */ |
| /* | - Ul_-8pl_tkl(..) | */ |
| /* database |  | * |
| /* |  | */ |
| /* BACKTracking : determ |  |  |
|  |  |  |
| utl_srt_prl ( [], []):- !. |  |  |
| utl_srt_prl( [Head\|Tail], Prioliet) : - |  |  |
| utl_spl_tkl ( Head, Tail, List1, List2), |  |  |
| utl_srt_prl ( List1, Listla), |  |  |
| utl_srt_prl( List2, List2b) |  |  |
| lst_apd lst ( Listla, [Head\|List2b], PrioList) |  |  |
| /* |  |  |
| /* PREDICATE | : utl_spl_tkl (...) | */ |
| /* |  | */ |
| /* |  |  |
|  |  |  |
| /* | TaskList1 | /* PARAMETERS : Task */ |
| /* | - TaskLista | */ |
| /* | - TaskList2 | */ |
| /* */ |  |  |
| /* Calls $\quad$ : SELF |  |  |
| 1* | [utl_ord_tsk(...) |  |
| /* - - |  |  |
| /* Database : -- |  |  |
| /* <br> /* BACKTRACKING . |  |  |
|  |  |  |
|  |  |  |
| utl_spl_tkl( $t k(T 2, E 2),[t k(T 1, E 1) \mid X],[t k(T 1, E 1) \mid Y], Z):-$ utl_ord_tsk( T2,T1), !, <br> utl_spl_tkl(tk(T2,E2), X, Y, Z). |  |  |
| $\begin{aligned} & \text { utl_spl_tkl( tk(T2,E2), } \operatorname{ttk}(T 1, E 1) \mid X], Y,\{t k(T 1, E 1) \mid Z]):-1, \\ & \text { utl_spl_tkl }(\operatorname{tk}(T 2, E 2), X, Y, Z) . \end{aligned}$ |  |  |
| utl_spl_tki ( _, [], [], []) |  |  |
| /* |  |  |
| /* PREDICATE : utl_ord_tak(...) |  |  |
|  |  |  |
| /* DESCRIPTION : Order <Task2>, <Task1> succeeds if <Task1> has higher lovel */ |  |  |
| /* PARAMETERS : Task2 */ |  |  |
| /* | Task1 | */ |
| /* |  |  |
| /* CALLS $\quad$ idb_tak_lvl(...) |  |  |
| /** ${ }^{\text {/* }}$ - */ |  |  |
| /* DATABASE |  |  |
|  |  | */ |

## D.4.11 des_utl.pro




NewLevel_Level + ExecTime,
utl_upd_lvi) Taskl, NewLevel
utl_lbl_all( LabelScheme, Ta






utl_est_exe ( B_lbl_hle, TaskName, Exectime): : !
idb_tek_exe( TaskName, Exectime).
idb_tek_oxel TaskName, ExecTime)





## D.4.12 des_wri.pro




 include "des_glb.pub"
include "des_cle.pub" include "des_cle.pub nclude "dcs_lst.pub"
nclude "dcs_trl.pub" include "dcs_wri.pub"
 /* Private predicate definitions $1 *========================================================================$ */ */




utl_rep_eat ().
utl_rep_eat ():-
utl rep_eat()


 wri_pre_chk( [Granteat, Fittoat, Crittest]):-








## Appendix E

## Original Publications

The research contained in this thesis has been reported both at international conference and invited elocution.

## E. 1 Conference Papers

Work described herein has resulted in the following conference publications:

- Linton K N, Terepin S, and Purvis A: "Parallel Digital Signal Processing for Audio Engineering", 88th Audio Engineering Society Convention, Montreux, Switzerland, March 1990, preprint 2917 (H-2).
- Linton K N, Gould G L, Terepin S, and Purvis A: "Optimising Massive Parallel Architectures for Real-Time Digital Audio", 89th Audio Engineering Society Convention, Los Angeles, USA, September 1990, preprint 2972 (F-II-6).
- Linton K N, Gould G L, Terepin S, and Purvis A: "Multiprocessor Architectures and Allocation Strategies for Digital Audio Mixing Consoles", Reproduced Sound 6, Windermere, Great Britain, November 1990.
- Linton K N, Terepin S, and Purvis A: "Task Scheduling Strategies for Digital Mixing Consoles", 90th Audio Engineering Society Convention, Paris, France, February 1991, preprint 3075 (G-3).
- Linton K N, Gould G L, Terepin S, and Purvis A: "Real-Time Multi-Channel Digital Audio Processing: Scalable Parallel Architectures and Taskforce Scheduling Strategies", 1991 International Conference on Acoustics Speech and Signal Processing, Toronto, Canada, May 1991, paper A2.7. ${ }^{\dagger}$

[^72]- Linton K N, Gould G L, Terepin S, and Purvis A: "On the Re-Allocation of Hardware Resources for Digital Audio Signal Processing", IEE Colloquium on Digital Audio Signal Processing, London, UK, May 1991, pp. 7/1-7/4.
- Gould G L, Linton K N, Terepin S, and Purvis A: "A Scalable Hybrid Multiprocessor for Real-Time Digital Audio Signal Processing", IEE Colloquium on Digital Audio Signal Processing, London, UK, May 1991, pp. 9/1-9/4.


## E. 2 Invited Papers

Following invitation, aspects of the research reported here have also been presented in:

- Linton K N: Real-Time Digital Audio Processing: Parallel Architectures and Resource Allocation Techniques, Invited Lecture to the British Section of the Audio Engineering Society, IBA, London, Great Britain, July 1991.
- Linton K N: "Architectural Issues for Parallel DSP Audio Systems", Invited Paper presented at the AES UK DSP Conference, London, September 1992, pp. 24-36. ${ }^{\dagger}$

[^73]
## Appendix $F$

## ICASSP '91 Paper Reprint

(See over for a reprint of the paper presented by the author at the IEEE International Conference on Acoustics Speech and Signal Processing, Toronto, Canada, May 1991).

# REAL-TIME MULTI-CHANNEL DIGITAL AUDIO PROCESSING: Scalable Parallel Architectures and Taskforce Scheduling Strategies 

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\# Solid State Logic Limited, Begbroke, Oxford, England, OXS 1RU


#### Abstract

In this paper we address the problems associated with the implementation of large-scale digital audio signal processing systems. For example, the audio mixing console requires the realisation of an architecture comprising in the order of 100 digital signal processors (DSPs). A hybrid multiprocessor, offering efficient computation with minimal communication overhead, is described and its suitability to the real-time environment discussed. Task allocation has a substantial impact on the efficient utilisation of such a DSP-engine. We are interested in algorithms that can produce optimal schedules in reasonable time and fast sub-optimal algorithms. An integrated software environment allows graphical input of DSP taskorces and hardware resource schematics, and the assessment of preferred scheduling strategies. We summarise the theoretical background, discuss the results obtained, and assess the implications of this research to the real-time audio processing environment.


## 1 Introduction

A number of multiprocessor architectures have been proposed in response to an ever-growing need for speeding up DSP-intensive applications. With the widespread availability of programmable DSP devices, a single scalable multiprocessor exhibits the ability to provide good performance for a range of real-time digital audio applications. One such application is the audio mixing console which, when implemented digitally, demands a computation engine rated in thousands of MIPS. With current technology, a topology comprising in the order of a 100 devices must be realised.

In order to efficiently develop such a system and allow unused processing power to be reallocated by the user in the operating environment, task scheduling strategies are necessary to efficiently distribute the set of processing algorithms required - termed the taskforce - across the architecture. A good scheduling algorithm will maximise and balance the utilisation of processing resources, and minimise the communication between processors. In addition any strategy must observe all precedeace relations, must be non-preemptive, and must not allow distributed execution of any task.

Sections 1 and 2 describe the preferred approaches to this problem, and their relative performance is ciscussed in Section 3. In Section 4 the architecture of the hybrid multiprocessor is detailed, with Sections 5 and 6 reviewing the data transfer techniques used. Finally, the current state of this research project and proposed further work is summarised.

## 2 Dynamic Programming

Dynamic Programming is employed in problems where finding a solution can be reduced to a sequence of interrelated decisions forming a path in a state-space. Scheduling taskforces on a multiprocessor architecture can be cast in these terms, as shown in Fig. 1. Each alternative decision from anode corresponds to scheduling a different task, or idle period, on a processor. At each step the non-intuitive option of scheduling an idle period, lasting until the next processor has completed its currently assigned task, must be allowed to guarantee optimality. Here the search is termed informed, as a set of altemative candidate nodes at each decision is kept, comesponding to the whole growing edge of the search tree.


Fig. 1: Task scheduling in terms of state-space searching.

The $A^{*}$ algorithm employs the best-first principle at each decision point, combined with problem-specific heuristics to draw the search process quickly towards a goal node, thereby avoiding aimless paths. Our implementation uses the minimum processor cost underestimate [1]. This corresponds to the optimistic estimate of the finishing time of the partial schedule, completed with all currently unassigned tasks. It is computed under the assumption that both the precedence relations, and the non-preemption and distributed execution constraints are relaxed.

## 3 Critical Path Method

In list scheduling strategies the tasks are first ordered according to some external prionity scheme, and then scheduled according to this regime. One technique used to form such a priority list is the Critical Path Method, originally investigated by Hu [2].

Although only optimal for taskforces consisting of equal duration tasks and exhibiting in-tree precedence constraints, the concept of Hu -level is widely applicable. A task $y_{i}$ is given the label $x_{i}$, where $x_{i}$ is the length of the longest path from $t_{i}$ to the root task, which is given the label ' 0 '. Tasks that are one unit removed from the root task are given the label ' 1 ', and so on as Fig. 2. The priority list is then ordered by non-increasing Hu -level, and the taskforce scheduled according to this regime.


Fig. 2: Hu-level labelling of a console taskforce.
Typically several root tasks are present, corresponding to a console taskforce with multiple outputs. In these cases, an artificial or "dummy" root task can be added to the taskforce. Tasks with execution times greater than one are represented by an indivisible chain of unit-weight tasks whose sum equals the weight of the original task. It can be shown that the level of the chain-task furthest from the root is the same as the level of the multi-unit task.

## 4 Scheduling Results

As a simple example, the $4: 1$ console taskforce of Fig. 3 requires 136 cycles of sequential processing, has an ideal schedule length of 62 cycles, and (ignoring bus delays) can be allocated optimally onto 4 processors as Fig, 4 and Table 1. Investigations with full channel strip processing enabled show that the $\mathrm{A}^{*}$ strategy results in considerable scheduling overhead when compared to the CPM approach. Furthermore, such large problem instances can result in extremely large search trees as the $A^{*}$ policy may degenerate towards an exhaustive search [3].

The level of performance achieved by the Hu-level Critical Path algorithm is typically well inside the theoretical performance bound of a factor of 2 from optimal. Results indicate that this approach produces such schedules in relatively short execution times. This can be attributed to the fact that in most cases console taskforces are configured as trees, or a forest of trees. In particular, the reader's attention is drawn to the allocation of a 16:8 console onto a eight processor architecture. Dynamics processing functionality was removed from four of the sixteen chamnels, allowing this $16: 8$ taskforce to be squeezed within the constraints imposed by the professional sampling rate [4].

In such large taskforces replacing the 2 -input mix tasks with 8 -input primitives, requining 10 instruction cycles, can greatly


Fig. 3: Example 4-input:1-output console taskforce.


Fig. 4: Optimal allocation of example taskforce

| Algorithro | Sequenfial Schesule | Parilld Schedule | Alger rithm Rentime | Speod-ap | Procassor Effidimies |
| :---: | :---: | :---: | :---: | :---: | :---: |
| $\mathrm{A}^{*}$ | 136 | 62 | 4:73 | 219.4\% | 59.7\% <br> $19.4 \%$ <br> $77.4 \%$ <br> $62.9 \%$ |
| Ho-hevel | 136 | 62 | 1:93 | 219.4\% | $\begin{aligned} & 39.7 \% \\ & 19.4 \% \\ & 71.4 \% \\ & 62.9 \% \end{aligned}$ |

Table 1: Performance assessments for above schedule.
reduce the scheduling overhead. This is due to two reasons: firstly, the number of tasks is considerably reduced, thereby significantly reducing the run-time of the $A^{*}$ and, to a lesser degree, CPM algorithm; secondly, the range of task size ratio is reduced by a factor of 2.6 , so reducing the number of $\mathrm{A}^{*}$ search-space nodes. However, this loss of relatively small tasks, and the tight packing previously available, means that high processor efficiencies are not as readily achieved.

## 5 Hybrid Hardware Module

The Inmos transputer is capable of parallel communication and computation, providing an ideal building block for the construction of scalable concurrent systems. However, the transputer is not effective at executing the data flow algorithms
required in real-time multi-chamel digital audio applications [5]. In contrast, DSPs such as the Motorola DSPS6001 are specifically designed to efficiently implement such algorithms [6] Due to inherent communication overheads, scalable architectures are not easily implemented with these devices alone. A hybrid multiprocessor assembled from nodes consisting of several Motorola DSP56001s hosted by a single transputer offers efficient computation with minimal communication overhead.


Fig. 5: Schematic diagram of hybrid processing node.
Inter-node communication is carried out using the transputer serial links, with one link reserved for broadcasting low bandwidth control data to the DSPs via a IMSC011 link adapter. Assuming a sampling frequency of 48 kHz and that block moves are used for data transfer, then each transputer is capable of supporting up to 34 audio paths in this trivalent configuration, Fig 6. Intra-node communication is achieved through the non-multiplexed extemal memory interface (EMI) of the IMST801 part. Each DSP possess an exclusive block of dual-ported RAM (DPR): the other port of the DPR is mapped into the transputer's EMI via appropriate decode Iogic.


Fig. 6: Temary tree configuration with hanging pipeline.

## 6 Intra-Node Communication

The transputer controls data transfer within the processing node. Data vectors pertaining to a particular DSP are externally placed by the transputer into the appropriate DPR. An arbitrary, and time varying, sofi configuration of the processors may be defined in software and mapped onto the hard configuration imposed by the hardware. Thus complex configurations, such as a hypercube or any random topology, may be realised with this system. Data integrity and synchronisation is assured by the use of semaphores [7]. The DPR is divided into domains which contain both input and output vectors, and an associated Boolean semaphore as Fig. 7. A processor will only effect a data transfer if the semaphore is in the correct state.


Fig. 7: Single domain DPR partitioning scheme.

Two of the major problems that arise in multi-processor systems are bus bottlenecking and spin-locking [8]. Bus bottlenecking is considerably reduced in this architecture as each DSPs bus is processor exclusive. Spin-locking occurs when a processor repeatedly tests and fails a semaphore, resulting in increased processor idle time. Although appropriate scheduling strategies can minimise spin-locking, a proposed extension to the single domain scheme promises to reduce the effects further. Here the DPR is partitioned into two domsins, each of which is accessed in a cyclic manner by both the transputer and the DSPs.

## 7 Semaphore Management

It is important to reduce the overhead associated with accessing and testing semaphores as much as possible. The DSP sees the semaphore as occupying a single word in external memory space and it is a simple matter to read a semaphore and to act upon its state. The approach taken with the transputer is somewhat different, however. Time critical sections of code are written in transputer assembly for optimum performance and embedded in an Occam2 hamess, as Fig. 8. The transputer uses a byte-oriented addressing scheme, beginning at the most negative address and working upwards. Instructions may be
stripped, and execution time minimised, by treating the semaphore as boolean values occupying contiguous byte locations in external memory space. This requires that two address decode schemes be implemented for the transputer.


Fig. 8: Occam 2 harness for dependent data paths.
The first, mapping contiguous blocks of DPR into contiguous words of transputer memory, is used for data areas. The second, mapping non-contiguous bytes contained in different DPR domains into contiguous bytes of transputer memory, for the semaphores. Due to the prefixing nature of transputer code, the semaphores may only occupy the sixteen locations beginning at byte location 0 to ensure zero prefixing. A restriction with this approach is that a maximum of sixteen semaphores may be used for maximum performance, and that additional address decoding needs to be utilised. It follows that a maximum of eight DSPs, using the twin domain approach, may be supported by a single transputer at full semaphore access speed.

## 8

## Conclusions

These encouraging results indicate that the sub-optimal strategy, based on the Hu-level Critical Path Method, consistently produces good-quality schedules with minimum overhead. Correspondingly this approach has direct application for the reallocation of unused processing resources by the engineer in the operating environment. In contrast, the optimal A* Dynamic Programming implementation may be utilised in the development cycle when assessing alternative multiprocessor designs, or in the production of pre-defined allocations yielding maximum system performance for a given architecture. Further results extend this concept by sacrificing channel processing to provide reverberation and other timedomain effects processing on a given hardware resource.

A target hybrid multiprocessor architecture has been described. The multiprocessor combines the efficiency of communication exhibited by the Inmos IMST801 transputer, and the computational power of the Motorola DSP56001. This novel architecture is fully scalable and capable of supporting additional processing elements whilst incurring negligible communications overhead. Further research will investigate the application of these strategies to architectures configured from the revolutionary Texas Instruments TMS320C40 part, and techniques for embedding resource re-allocation methods on such platforms [9].

## Acknowiedgements

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## Appendix $G$

## AES UK DSP Paper Reprint

(See over for a reprint of the invited paper presented by the author at the Audio Engineering Society UK DSP Conference, London, Great Britain, September 1992).

# ARCHITECTURAL ISSUES FOR PARALLEL DSP AUDIO SYSTEMS 

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#### Abstract

The design and implementation of audio systems through analogue technology is a well understood art. With the success of both professional and consumer digital formats, the audio industry is moving quickly towards an entirely digital signal path. In this paper, the principle architectural issues of implementing digital audio systems with general purpose digital signal processors are discussed. Particular attention is given to the design and performance aspects appropriate to professional audio products.


## INTRODUCTION

Digital signal processing (DSP) devices are inherently programmable. Instead of using different types of hardware to accomplish different processing functions, DSPs just perform different sets of summations, delays and multiplies grouped together as processing algorithms. With a DSPbased audio system, the limiting factor is the total amount of processing power available rather than the quantity and type of processing hardware provided. The modularity, flexibility, and reliability of DSP makes it attractive for all manner of applications, including those traditionally implemented by analogue technology.

The advent of relatively low cost DSPs has made multi-DSP processing an economic reality in today's digital audio environment. As the audio community becomes accustomed to buying more technology with less money, design engineers face increasingly difficult design problems. In most cases, this requires that the designer opt for a software-based multiprocessor (M-P) system built from freely available off-the-shelf components from a range of manufacturers. This strategy allows a quicker design cycle and a flexible approach to the design of professional audio products.

## 1 DESIGN CONSTRAINTS

In the design of parallel DSP audio systems, the first three constraints are those inherent in conventional uni-processor designs: processor speed, memory resource and input/output (I/O)
bandwidth. Since most applications require more processing power than is provided by a single DSP device, a multi-DSP architecture must be created. Memory capacity is not usually of concern since DSPs can address much more memory than can be accessed in a sample period. As most DSPs offer independent $I / O$, supporting $A D C / D A C s$ and AES-EBU/SPDIF interfaces, multi-DSP M-Ps offer excellent I/O bandwidth which scales with the number of processors.

### 1.1 Initial Decisions

The two most fundamental decisions which must be taken very early on in the design process are, firstly, how powerful each DSP should be, and, secondly, how many processors should be supported. For a system of required performance $P$, and an ideal architecture containing $n$ processors, each with an individual processing capability of $p$, the hyperbolic relationship between $n$ and $p$, as Figure 1, defines a span of possible architectures satisfying [1]. Therefore, one could use a small number of very powerful (and expensive) processors, or a. large number of relatively slow (and cheap) processors. The use of large numbers of DSPs is made attractive by the development of relatively cheap DSP devices suitable for professional audio applications.
The cost advantage of using low-speed technology is balanced by the degradation in efficiency that inevitably occurs as the number of processors in a system increases. Moreover, the complexity of programming a M-P with many DSPs far exceeds the complexity of programming an architecture
consisting of just a few processors. Consequently, although economics might enhance the attractiveness of a system with many low-speed DSPs, the advantages disappear if efficiency is not held high. When the parallelism cannot be tapped effectively, it simply adds to system cost and complexity. This demands that multi-DSP systems are assembled in such a way that the performance of each DSP is effectively hamessed.

### 1.2 Implementing Parallelism

There are two fundamental ways in which DSPs can be composed to create parallel architectures. Perhaps the simplest way to introduce parallelism is to replicate a component $n$ times, as shown in Figure 2a. To exploit this form of parallelism, the digital audio streams processed must be separated in space into individual channels. For this reason this form of parallelism is known as spatial parallelism. A typical example from everyday life is the familiar row of checkout desks in a supermarket.

The other fundamental way of introducing parallelism is to partition the audio processing into a number of steps, as shown in Figure 2b, which when applied sequentially to each sample perform the original processing desired. In other words, the task is partitioned in time, with each step of the application being applied to a separate sample. For this reason this form of parallelism is known as temporal parallelism. The application of temporal parallelism in digital audio produces pipelined structures.

### 1.3 Processor Pipelining

Pipelining is an implementation technique whereby multiple tasks are simultaneously overlapped in time [2]. In fact, pipelining is the key implementation technique used to make fast DSP devices. In a processor pipeline, as in an assembly line, the work required to complete a processing application is broken into smaller pieces or stages. The stages are then connected one to the next to form a pipe, as Figure 3a. Samples enter at one end, are processed through the stages, and exit at the other end, as detailed in Figure 3b.
In digital audio systems response is perceived as intimate if the system operates with a latency, or accumulated processing delay, of less than one millisecond. To meet the necessary frequency response requirements, each DSP in the audio system must take a new set of input samples every $20.8 \mu \mathrm{~s}$, given the professional sampling rate of 48 kHz . This corresponds to a maximum processor
pipeline of 48 stages between any input and any output.

### 1.4 Additional Constraints

Three additional design constraints, which are not usually associated with uni-processor systems, stem from the need for parallel architectures. Firstly, with multiple DSPs it becomes necessary to exchange audio samples and control parameters so some type of interprocessor communications (IPC) network is required. Novel techniques are required to tie together conventional DSPs in large digital audio systems. Secondly, because the processors must work together they are required to synchronize to coordinate such activities as access to shared data. Finally, cost-effective programmable DSP systems necessitate M-P architectures which incorporate sufficient flexibility in their design to support several different audio processing applications.

## 2 M-P CONFIGURATIONS

Most professional audio applications require a processing rate much higher than can be provided by any single DSP device. For example, a 2 channel third-octave graphic equaliser operating at the professional sample rate ( 48 kHz ) requires about three 10.25 MIPS DSPs. A zero chip interface may be implemented by connecting the serial links of two or more DSPs. Unfortunately, the relative slowness of this technique restricts this approach to a limited number of applications. For faster data transfer, the parallel data bus must be used to connect multiple DSPs in tightly coupled M-P architectures.

Conceptually, the simplest way to interconnect multiple DSPs is to provide full connectivity between processors. However, due to the sheer hardware costs involved, designers must forgo this luxury. Shared-memory architectures are probably the most common form of M-P, since they are the natural extension of conventional uni-processor systems. A radically different communication mechanism is message-passing. In messagepassing architectures DSPs communicate by sending and receiving messages via dedicated communication channels [1].

### 2.1 Shared-Memory Design

The simplest way to construct an audio M-P is to connect the DSPs on a shared bus, providing shared common memory to all processors, as shown in Figure 4. In order to isolate the processors from each other so that they can
simultaneously access their local memories, bidirectional bus transceivers are required. The bus is then active only when data is transferred between DSPs, allowing the bus to remain free during processing which does not demand IPC. If the local DSP RAMs were not present, there would be severe contention on the bus, causing arbitration delays to reduce performance.
Unfortunately, in the case of processors such as the DSP56001 there is no provision to force the external memory port to quickly tri-state (isolate) in a deterministic manner. Since the address bus is always driven (except by granting the bus), one DSP's local RAM cannot be directly accessed by another DSP while the first accesses its internal RAM. Additional bus transceivers would be required to allow one DSP to quickly access another DSP's local RAM without shorting its main bus, increasing board real-estate and power consumption [3].

### 2.2 Arbitration Schemes

Arbitration schemes are necessary to resolve the situation when multiple processors need to access the same piece of data at the same time. Most DSP devices have provision for asynchronous sharing of their external memory port, with the use of busrequest (BR) and bus-grant (BG) signals. M-P architectures which use BR and BG signals to directly share a common bus typically require a hardwired master-slave relationship between the interconnected DSPs. This situation is certainly suitable for many product designs. However, software control is to be preferred, as this provides an equal relationship between the processors.
A second disadvantage of this technique is that the process of granting the bus is too time consuming for most real-time audio applications, as there is little time in one 48 kHz sample period for arbitration of the bus. In synchronous designs, the most serious problem with this technique is the indeterminate timing of the bus-granting process. For example, the BR to BG delay for a 20.5 MHz (10.25 MIPS) DSP56001 is specified to take between 73 to 240 ns for zero-wait states [4]. Since this delay may be more than one instruction cycle, synchronous DSPs cannot be kept in step when sharing a common bus via the $\mathrm{BR} / \mathrm{BG}$ arbitration technique.

### 2.3 Summary

An $n$-processor system requires a bus whose bandwidth is of the order of $n$-times that of a uniprocessor bus. Consequently, the major
shortcoming of shared-memory M-Ps is selfevident: the data transfer capacity between processors is determined by the bandwidth of the bus and is therefore constant. As the number of DSPs grows, the shared bus between DSP modules becomes a serious bottle-neck to IPC. This factor limits the number of processors that can be usefully incorporated into such a system, and hence fixes an upper limit on extensibility.
Typically, bus-oriented systems can support 10 DSPs efficiently and possibly stretch to 20 or 30 . Beyond this range, bus contention leads to degraded performance to the extent that such systems are unlikely to support large audio systems unless a technological breakthrough provides a very high bandwidth bus at relatively low cost. A possibility for the future is to implement large digital audio systems with optic fibre links. However, the low cost and simplicity of busstructured M-Ps has proved to be advantageous for a number of small-scale parallel DSP systems, yielding high performance/cost ratios without any pretensions of scalability.

## 3 SYNCHRONIZATION MECHANISMS

Multiple DSPs can be synchronized at the instruction cycle level through the use of a common clock. However, clock skew can be a considerable problem especially at high clock rates in large M-P architectures. Alternatively, synchronization mechanisms are used to ensure that shared data is always valid in asynchronous designs. This is one of the most difficult and errorprone types of DSP programming that exists, because it involves the understanding of the potential simultaneous actions of multiple DSPs.

### 3.1 Semaphores

The dictionary defines semaphore as "signalling by flags". Semaphore flags are used like locks to shared resources, such as hard disk buffers and interprocessor message queues, and so permit only one DSP at a time to gain access. Two different operations are performed on the semaphore: the request operation which attempts to gain access and the release operation which signals the termination of the access. These operations are used to guarantee mutual exclusion, meaning that only one processor is able to access shared data at any given time, locking out all other processors. This occurs from the time a request is granted until the time that the semaphore is released.

At one extreme a whole shared memory can be controlled by a single semaphore, whereas at the
other extreme every data item could be individually locked. The former precludes any parallel operations on the shared data, of course, while in the latter case the overhead of performing the locking operations would be prohibitive. Typically, this overhead is distributed across several data values, with one semaphore locking the corresponding data structure.

### 3.2 Test-and-Set

One of the most basic synchronizing techniques is the test-and-set primitive [5]. Firstly, the DSP desiring to obtain exclusive access to the shared data, reads the corresponding semaphore flag. It then tests this prior semaphore value to determine if it was successful (i.e. a 0 ) and, finally, writes a 1 to the semaphore location. As a result, the test-andset operation forces the semaphore flag to be set to 1 , whether or not it was set beforehand. If the semaphore was not already set, then this DSP can access the shared resource. All other DSPs are blocked because the semaphore is now set. As a result, only one DSP at a time has permission via the semaphore.

To ensure that it can be used successfully, the test-and-set operation must be uninterruptible. That is, once it is initiated and the read access is completed, no other access can be made to the semaphore until the semaphore is rewritten during the second step of the test-and-set. If an intervening access were permitted, synchronization could fail. This is because one DSP may test the semaphore and, before it can set it, another DSP might test it. In this case, both processors believe they have the semaphore. Most DSPs support an indivisible READ/MODIFY/WRITE instruction, which ensures that the address bus remains active throughout the entire operation.

If several DSPs execute a test-and-set on a semaphore concurrently, the requests will be serialised and executed one by one due to the indivisible nature of the READ/MODIFY/WRITE operation. Given serial execution, no more than one DSP from a set of concurrent requesters can observe a 0 semaphore value and thereby gain access to the shared resource. When it has finished with the resource the processor must release the semaphore. To achieve this, the owner DSP does no more than clear the flag by writing a 0 into the semaphore location. Note, it is not necessary to perform a READ/MODIFY/WRITE instruction to unlock the semaphore.

### 3.3 Spin-Lock

The test-and-set is only half of the synchronisation mechanism: the other is the action taken depending on whether the semaphore has been granted or not. A question which immediately arises is what to do with a DSP which is waiting to access a shared resource. One mechanism which gives fast entry to a shared resource is the spin-lock. Thus, if the lock has been granted, the processor continues on to use the shared resource. If not, then a spin-lock occurs, and the DSP must spin backwards and re-execute the test-and-set, repeating the process until the lock is granted, as Figure 5. There are two major drawbacks to using spin-locks [6].

Firstly, a DSP which is polling a semaphore is not doing any useful work. Secondly, it is consuming memory cycles in the bank containing the semaphore value and, if several processors are waiting at the same semaphore, memory bandwidth is rapidly consumed. This contention causes additional cycles of delay while a DSP is attempting to release a lock, which magnifies the effect of the bottle-neck at the semaphore. As a result, spin-locks waste valuable MIPS by dedicating potentially useful DSP instruction cycles to the effort of repeatedly testing semaphores.

## 4 PERFORMANCE ISSUES

Characterising the performance of DSP engines is an important exercise if designs are to be compared, particularly with respect to scalability. A realistic assessment must take into account both the architecture and the applications for which it is intended. Characterising a particular audio application includes considering processing, memory, I/O, and IPC resources required. Similarly, proposed architectures should be considered in terms of the hardware resources available to the range of applications being considered [7].

### 4.1 MIPS

A number of popular measures have been adopted in the quest for a standard measure of M-P performance. Small, key pieces extracted from real audio applications, such as biquad filter algorithms, can provide useful processing kernels for performance evaluation. One alternative to time is the parameter MIPS million instructions per second. For example, Table 1 details a hypothetical processing budget for a comprehensive digital mixing console [8]. Since

MIPS is the rate of operations per unit time, faster architectures have a higher MIPS rating.
Although MIPS is easy to understand, the problems with using MIPS as a measure for comparison are two-fold. Firstly, MIPS is dependent on instruction set, making it difficult to compare M-P systems based on DSPs with different instruction sets. Secondly, MIPS can vary widely between different processing applications on the same audio system. As a result, the only consistent and reliable measure of the performance of M-P systems for a particular audio product is the actual execution time of the required processing algorithms.

### 4.2 Efficiency

In M-P systems efficiency is defined in terms of the ratio of time spent executing useful DSP routines compared to that consumed by IPC, synchronisation and other overheads. When a M-P is operating at peak performance, all processors are engaged in useful work. Moreover, no processor is executing an instruction that would not be executed if the same application was executing on a single DSP. In this state, all $n$ processors are contributing to effective performance, and the MIPS is increased by a factor of $n$ - linear speedup. Unfortunately, this is rarely achieved due to the overheads incurred as a direct result of the use of parallelism.
In one case, designers started out with the knowledge that in previous M-Ps each additional DSP contributed only 0.8 times the actual performance of each processor already in the system [1]. Hence, points on the speed-up curve for such a system would typically be $1,1.8,2.5, \ldots$, as shown in the 'actual' curve of Figure 6. Such diminishing returns make each successive processor less and less cost effective. Efficiency is clearly a major concern in the design of digital audio products based on M-P architectures: a design that uses $2 n$ DSPs inefficiently cannot compete on a cost basis with a design that uses $n$ identical processors twice as efficiently.

### 4.3 Load Balancing

There are various options when executing an audio processing application consisting of $m$ processing algorithms on an $n$ processor multi-DSP system. If all the work is allocated to one DSP and the remaining processors ignored, the IPC overhead is minimised but the system will perform no better than a single DSP. Conversely, if the work is split between the $n$ processors, considerable overhead is
incurred. Here, performance could be up to $n$-times that of a single DSP device, as shown in the ideal curve of Figure 6. Experiments with M-P architectures, however, have demonstrated behaviour in operational systems is more like the 'actual' curve.

In practice, the speed-up increases significantly only for the first few additional processors. At some point the speed-up may actually begin to decrease with each additional DSP this can be as few as 10 depending on the architecture. This decrease in expected speed-up is due to excessive IPC and synchronization overheads, incurred when processing algorithms resident on different DSPs must communicate with each other. The resultant degradation in performance for incremental increase in hardware resources is known as the saturation effect. Consequently, the efficiency of a system depends critically on how the workload is balanced between the processors.

## 5 FURTHER CONSIDERATIONS

There are many ways to express the performance of a DSP system. For example, the metric of performance commonly assumed is MIPS, but cost, reliability and extensibility are just as important.

### 5.1 Cost

An understanding of cost is essential for designers to be able to make intelligent decisions about whether or not a particular processor, I/O or interconnection component should be incorporated in design. The cost of DSP components is changing so fast that good designers are basing design decisions not on today's costs, but on projected costs at the time the product is shipped. Although they are far from representing what the customer must pay, cost of components confine a designer's desires. In digital audio systems, component costs typically make up $15-33 \%$ of the final list price.
Many engineers are surprised to find that most companies spend only $8-15 \%$ of their income on R\&D. This information suggests that a company uniformly applies fixed-overhead percentages to turn cost into price. Another point of view is that R\&D should be considered an investment, and so an investment of $8 \%$ to $15 \%$ of income means that every pound spent on $\mathrm{R} \& D$ must generate $£ 7$ to $£ 13$ in sales. This alternative point of view then suggests different gross margins for each product depending on the number sold and the size of the investment [2].

Large expensive DSP. systems generally cost more to develop - a product costing 10 times as much to manufacture may cost many times as much to develop. Since large audio systems generally do not sell as well as small ones, the gross margin must be greater on big machines for the company to maintain a profitable return on $R \& D$ investment. This places large DSP systems in double jeopardy, since there are fewer sold and they require larger R\&D costs.

Few customers can afford the initial cost of a large multi-DSP system. However, many can afford to grow a large system in increments as they discover the need for more processing power and as their budgets permit. There is also a great risk in designing a multi-DSP architecture welded to a particular audio processing application. By incorporating sufficient flexibility into a design to cover several applications, the $\mathrm{R} \& \mathrm{D}$ investment can be amortised over a range of end-products constructed from affordable DSP modules.

### 5.2 Reliability

The reliability of M-P systems is often considered to be an issue which is of secondary importance to the task of designing for maximum throughput. It is a commonly held belief that M-P architectures are inherently tolerant of faults since the replication of processing elements leads to the natural availability of spares. Designing machines which are fault-tolerant (and hence reliable) involves a great deal more that simply providing spares, however. Each fault must be located before any hardware re-configuration can be performed.
As in any electronic system, the most unreliable elements in a multi-DSP architecture are the electrical connections between physically distinct component parts. Since interconnection networks normally contain large numbers of wires and connectors, intermittent faults caused by momentary interruptions are not uncommon. Therefore the protocol for data-movement through the network should be robust and capable of detecting errors. More permanent faults can result in one or more DSPs being unable to communicate with the rest of the M-P.
Since faults cannot be avoided, the only alternative is to design multi-DSP systems for maximum resilience and fault-tolerance. Designing for maximum resilience means discovering which components are least reliable, and either minimising their use or making them more reliable. Designing for fault-tolerance means two things: firstly, designing systems with the ability to
detect the occurrence of an error, and secondly imbuing those systems with the ability to correct and recover from an error.

### 5.3 Extensibility

It is important to leave room for future expansion of an audio M-P as one runs out of processing power. For example, a shared bus would typically experience heavy traffic with only 10 DSP modules. Many professional audio applications require much more processing power than this. Therefore, a fundamental consideration is the effect on the system of altering the degree of replication, usually upward. When assessing the extensibility of a particular M-P machine, a designer must consider several aspects of the design.
Firstly, as the number of DSP modules is increased the total cost of the system must also increase. It is obviously desirable to minimise this increase in cost to that of linear growth. Secondly, the designer must assess how the basic architecture parameters, such as IPC capacity, vary with replication. A further important practical consideration, especially for very large scale M-Ps, is the space occupied by the system as a whole and, more particularly, by the interprocessor wiring.
If the degree of replication is to be very large then a careful analysis of the rate of growth in hardware complexity must be performed. However, the ability to extend a particular multi-DSP architecture is clearly only important within a given range, since commercial systems have a finite lifetime and end-users have finite budgets. In the longer term, as configurations become larger, the effects of scaling will become more important. The extensibility of M-P architectures is still a subject which is of much research interest.

## 6 MESSAGE-PASSING DESIGN

Various interconnection schemes have been suggested for message-passing between conventional DSP processors. Here, some of the approaches which utilise widely available integrated devices are considered.

### 6.1 First-In First-Out Buffers

One technique is to employ first-in first-out (FIFO) buffers. It is important to monitor the boundary conditions (FULL or EMPTY) of these devices, as failure to act on their occurrence will result in data overflow or underflow. The current FIFO generation signals the EMPTY, HALF-FULL and FULL conditions by asserting corresponding
external pins. The EMPTY and FULL flags are also fed back internally and inhibit further read and write until the FIFO is no longer empty or full.
The increasing use of high-speed applications has created a demand for a faster and smarter generation of FIFOs. Flagged FIFOs offer the basic features discussed above while providing two additional programmable flags: ALMOSTEMPTY and ALMOST-FULL. These flags can be used as early warning flags in critical real-time applications such as pipeline DSP for digital audio applications. Current devices provide for unlimited expansion while maintaining a 50 ns fall-through time [9].

### 6.2 Dual-Ported RAM

Due to their high bandwidth and message passing flexibility, dual-ported RAMs (DPRs) can be used to link multiple high-performance DSPs. Several manufacturers produce DPRs in many configurations, all of which operate at static RAM speeds ( $50-150 \mathrm{~ns}$ ) and have two independent external memory ports [9]. This allows two processors to share the same block of physical memory in their respective address spaces. The two DSPs can access data in two memory locations simultaneously and asynchronously.

This approach clearly out-performs a discrete parts design where two processors must synchronize through arbitration for access to a bus which is used to access one location at a time in a standard single-port RAM. In this way, the integrated DPR approach removes synchronization requirements at the memory's bus access level. Nevertheless, synchronization must be performed at other levels to ensure data integrity and thus proper system operation.

### 6.2.1 Busy Logic

A problem can occur with DPR memories when both ports attempt to access the same address at the same time. There are two significant cases: when one port is trying to read the same data that the other port is writing and, when both ports attempt to write to the same word at the same time. If one port is reading while the other port is writing, the data on the read side will be changing during the read and a read error can result. If both ports attempt to write at the same time, the memory cell is being driven by both sides and the result can be a random combination of both data words.

Busy logic solves this problem by detecting when both sides are using the same location at the same time. The BUSY output pins are suitable for
attachment to the WAIT inputs of most DSPs. This approach is very straightforward and flexible, and has the benefit that a processor cannot be locked out of the RAM longer than the access period of the other processor. Note that although one or the other processor may have to wait occasionally, the throughput loss is minimal since the probability of both DSPs using the same location at the same time is very small.

### 6.2.2 Semaphore Support

Software constraints may require mutual exclusion at the data structure level rather than at the memory location level. Instead of comparing addresses on every cycle and occasionally asserting BUSY, other DPR devices employ additional circuitry to support semaphores, thus ensuring that only one side has permission to use a block of memory. The conventional test-and-set semaphore operation requires that the two memory accesses are indivisible: some DPR devices employ a twist by using set-and-test.

The set corresponds to a request and the test checks to see if the request was granted. The indivisible double access requirement is avoided because, as soon as the request is made by one DSP, the grant is blocked on the other side. As a result, semaphores can be supported for DSPs which do not provide an indivisible READ/MODIFY/WRITE instruction. Since there is no hardware relationship between semaphores and DPR locations, the block sizes and locations, and semaphore association can be defined in software. This offers the system designer considerable flexibility.

### 6.2.3 DPR Memory Expansion

DPR chips can be combined to form large dualport memories. Expansion in memory depth with DPRs is similar to that used for conventional RAMs, where the top address bits enable individual memory blocks. DPRs can also be expanded in width. However, in this case, we have a subtle problem. If the addresses for both ports arrive simultaneously at both RAMs, it is possible for both sides to simultaneously assert BUSY, causing both DSPs to wait indefinitely for their ports to become free. The solution to this BUSY lock-up problem is to use the arbitration logic in only one DPR.
An example of expanded DPRs used for DSP-toDSP communication is shown in Figure 7. Here, two DSP56001s communicate using three 8-bit wide DPR chips in a MASTER/SLAVE
configuration to provide a word-wide dual-ported memory. If the MASTER device activates BUSY, the dedicated DPR SLAVE chips, which incorporate a BUSY input pin, will internally disable their write enables. Note that the BUSY signals from the MASTER are also fed to the WAIT pins of the DSPs, forcing them to stall when their side of the DPR block is BUSY.

### 6.2.4 DPR and FIFOs Compared

Each $2 \mathrm{~K} \times 24$-bit DPR block provided for interDSP communication requires three 48 -pin ICs, plus components for memory decoding. A bidirectional $2 \mathrm{~K} \times 24$-bit FIFO channel requires a total of six 28-pin ICs. With each doubling in size of a DPR block, 2 address pins must be added to the chip package. Larger FIFOs never need to change their package size, since the cells within the FIFO are not directly addressed. Thus FIFO technology scales better than comparable DPR technology. However, DPRs provide considerably more communications flexibility.

### 6.3 Novel Devices

Serving as a complex four bus interconnection network, quad-ported RAMs (QPRs) greatly simplify the task of creating generalised M-P systems for audio processing with conventional DSPs. Although typically three times as expensive as comparable DPR devices, QPRs have a considerable interconnectivity advantage over DPRs. For example, in order to fully interconnect four DSPs requires 1 bank of QPR as opposed to 6 banks of DPR, as shown in Figure 8.

Implementing multi-DSP systems with processors not specifically designed for parallel processing can cause IPC to quickly saturate device I/O and adversely affect processing efficiency. DSP manufacturers made the first step in addressing the need for parallel processing by providing designers with two external memory interfaces. Novel devices such as the Texas Instruments TMS320C40 go several steps further by incorporating on-chip hardware to support dedicated communications ports [10].
Each of the six bidirectional communication ports provides glueless processor-to-processor communication at a transfer rate of $20 \mathrm{Mbytes} / \mathrm{s}$. Without these ports, the corresponding $120 \mathrm{Mbytes} / \mathrm{s}$ of processor throughput would have to be squeezed into one or both of the external memory interfaces, as Figure 9. With the communication ports, IPC bandwidth is plentiful and scales with the number of processors. In this
way the TMS320C40 can support of a wide variety of complex M-P configurations to meet the needs of real-time audio processing applications.

## 7 CONCLUSIONS

Parallel DSP systems are an economic reality in today's digital audio environment, due to the wide availability of low cost DSPs devices. The fundamental advantage of such multi-DSP designs is their generality, since software algorithms govern the manipulation of samples and so dictate the audio processing performed. Although it is easy to replicate the processing resources required for multi-DSP audio systems, such replication can introduce serious problems with respect to communications bandwidth.

IPC networks are of fundamental importance in parallel DSP systems as they directly effect system throughput and have a bearing on modularity, extensibility and overall system performance. In order to implement digital audio systems through such replication, interconnection networks are required which provide sufficient IPC bandwidth for the range of processing applications being planned.
Utilising design techniques from conventional uniprocessor systems allows small M-P systems to communicate through a common bus and shared memory. However, the inherent advantages of low cost and simplicity must be balanced against the significant problems encountered when this approach is used for large-scale audio systems.
Integrated interconnection components, such as FIFOs and multi-ported RAMs, provide simple static RAM interfaces which connect transparently to almost any existing DSP design. As a result, they enable conventional DSP modules designed for uni-processor or small shared-bus systems to be directly employed in a variety of scalable M-P architectures. In particular, the multi-ported RAM scheme presents a cost effective method for message-passing between DSP modules.
Knowledge accumulated through the research of the past 25 years into parallel architectures and programming is making considerable impact on the design of products for the digital audio environment. For example, software techniques, such as semaphore flags, guarantee the integrity of audio samples shared by multiple DSPs. Task allocation strategies minimise the effect of IPC saturation and maximise the processing efficiency of multi-DSP systems [7].

Novel devices, such as the Texas Instruments TMS320C40, are beginning to emerge which integrate multiple IPC ports directly on-chip with high-speed DSP cores. Although currently expensive, their cost compares favourably with the pricing of conventional DSPs, such as the Motorola DSP56001, when they were first released.

The design of multi-DSP audio products is extremely challenging: one cannot simply lash together 100 DSP devices and expect to obtain 100 -times the performance of a sole DSP. Although the amount of raw processing power is important for audio processing applications, factors such as efficiency and cost are of considerable consequence in the marketplace. Understanding of the architectural approaches available and the implications of design decisions enables the construction of well-engineered parallel DSP audio systems.

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Figure 1: Individual DSP Power against number of DSPs required


Figure 2a: Exploiting spatial parallelism


Figure 2b: Exploiting temporal parallelism


Figure 3a: Processor pipelining with DSPs: four-stage linear pipeline


Figure 3b: Detail of overlapped processing in 4-stage linear pipeline


Figure 4: Example shared-memory multiprocessor architecture

| Digital Mixing Console Design | Processing Budget |
| :---: | :---: |
| No. of functions for each channel strip | 80 discrete and 30 continuous |
| No. of functions for complete 64 channel console | 7000 functions |
| No. of functions at professional sample rate of 48 kHz | 350 million functions/s |
| Processing requirement, at 3 instructions per function | $>1000 \mathrm{MIPS}$ |
| Size of M-P architecture, using 10 MIPS DSPs | $>100$ processors |

Table 1: Processing budget for a hypothetical digital mixing console


Figure 5: Flow chart of test-and-set and spin-lock semaphore operations


Figure 6: Speed-up curves for multi-DSP architectures


Figure 7: DSP-to-DSP communications using width-expanded dual-ported RAMs


Figure 8: Using quad-ported RAM to fully interconnect conventional DSP processors


Figure 9: Advantages of using dedicated communications ports on TMS320C40

## Appendix H

## Glossary of Abbreviations

A glossary of abbreviations used in this thesis is given below. Abbreviations which appear in only one section may not be listed here, but are always defined when they are first used.

| A-D | : analogue-to-digital |
| :--- | :--- |
| ADC | : analogue to digital conversion (or converter, depending on context) |
| ADT | : automatic double-tracking |
| AES | : Audio Engineering Society |
| ALL | : all task-processor pairs/heuristics - DYN/HRT control code used in DCS |
| ALU | : arithmetic logic unit |
| ANSI | : American National Standards Institute |
| AOT | : adjusted on test |
| ASIC | : application specific integrated circuit |
| ASP | : audio signal processing |
| AST | : A* dynamic programming |
| BG | : bus grant |
| BR | : bus request |
| CD | : compact disc |
| CO | : combinatorial optimisation |
| CMOS | : complementary metal oxide semiconductor |
| CPM | : critical-path method |
| CPU | : central processing unit |


| D-A | digital-to-analogue |
| :---: | :---: |
| DAC | : digital-to-analogue conversion (or converter, depending on context) |
| DLL | dynamic-link library |
| DMA | : direct memory access |
| DMC | digital mixing console |
| DPR | : dual-ported RAM |
| DSP | digital signal processing (or processor, depending on context) |
| DYN | dynamic-level list scheduling |
| EBU | European Broadcasting Union |
| EIAJ | (Japanese Electronics Manufacturers' Association). |
| EQ | equalisation |
| EMI | : external memory interface |
| EVB | : evaluation board |
| FET | : field effect transistor |
| FFT | : fast Fourier transform |
| FIFO | : first-in first-out |
| FIR | : finite impulse response |
| FM | : frequency modulation |
| FWE | : fully-overlapped with execution |
| GA | : genetic algorithm |
| G-P | general-purpose |
| H-F | : high-frequency |
| HLE | : highest level first, estimated times |
| HLN | : highest level first, no estimated times |
| HMF | : high mid-frequency |


| H-P | : high-pass |
| :--- | :--- |
| HRTF | $:$ head-related transfer-function |
|  |  |
| I/O | : input/output |
| IEEE | $:$ Institute of Electrical and Electronic Engineers |
| IID | $:$ inter-aural intensity difference |
| IIR | : infinite impulse response |
| IN | $:$ interconnection network |
| IPC | $:$ inter-processor communication |
| ISO | $:$ International Standards Organisation |
| ISPW | $:$ IRCAM Sigual Processing Workstation |
| ITD | $:$ inter-aural time delay |

LC : link connectivity
LCE : lowest co-level first, estimated times
LCN : lowest co-level first, no estimated times
L-P : low-pass
LCD : liquid crystal display
LED : light emitting diode
L-F : low-frequency
LFO : low-frequency oscillator
LGDF : large-grain data-flow
LMF : low mid-frequency
$\mu \mathrm{P} \quad$ : micro-processor
MAC : multiply accumulate
MADI : multi-channel audio digital interface
MFLOPS : million floating-point operations per second
MIA : minimum independent assigument

| MIDI | : musical instrument digital interface |
| :---: | :---: |
| MIMD | : multiple-instruction multiple-data |
| MIPS | : million instructions per second |
| MISD | multiple-instruction single-data |
| MMI | : man-machine interface |
| M-P | : multi-processor |
| MPC | : minimum processor cost |
| MRC | : maximum remaining critical-path |
| MTBF | : mean time between failures |
| NC | : node connectivity |
| NML | : non-overlapped, multiple links |
| NMOS | : n-type metal oxide semiconductor |
| NON | : none - DYN/HRT control code used in DCS |
| $N P$ | : non-deterministic polynomial-time |
| NSL | : non-overlapped, single link |
| $P$ | : polynomial-time |
| PAM | pulse-amplitude modulation |
| PC | : program counter |
| PCM | : pulse-code modulation |
| PE | : processing element |
| PFL | : pre-fader listen |
| PPM | peak programme meter |
| PRO | : processor with lowest index |
| PWM | : pulse-width modulation |
| QPR | : quad-ported RAM |

R\&D : research and development
RAM : random access memory
RISC : reduced instruction set computer
RND : random static labelling
ROM : read only memory
RSS : Roland Surround Sound

SA : simulated annealing
SCFET : smallest co-level first, estimated times
SCFNET : smallest co-level first, no estimated times
SCI : serial communications interface
SIMD : single-instruction multiple-data
SISD : single-instruction single-data
SMPTE : Society of Motion Picture and Television Engineers
SNR : signal-to-noise ratio
SP-DIF : Sony/Phillips Digital Interface Format
SPW : Signal Processing Worksystem
SQNR : signal-to-quantisation-noise ratio
SSI : synchronous serial interface
SU : super-unitary

TDA : total dynamic automation
TDM : time-division multiplexing
TI : Texas Instruments
TLB : translation lookaside buffer
TRC : Torus Routing Chip
TSK : task with lowest static level
TSP : travelling salesman problem

VCA : voltage-controlled amplifier
VDU : visual display unit
VHDL : VLSI hardware description language
VLSI : very large-scale integration
VU : volume unit

XLL : Excel add-in DLL

## Appendix I

## Bibliography

While not explicitly referenced within the main text, several books contain subject matter of particular relevance to the research reported in this thesis. Titles in this category include:
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## Colophon

This thesis was written, illustrated and published using Frame Technology 's FrameMaker on NeXT and IBM-PC compatible workstations. Analytical results from both $T P G$ and $D C S$ deliverables were analysed and charted using Microsoft Excel. Additional graphics were generated using Mathematica, from Wolfram Research, running under NeXTstep.

The TPG C source included in Appendix C was developed using Microsoft's Visual C++ IDE and built as an XLL dynamic-link library for Excel. Borland's Turbo Prolog compiler was used to develop the $D C S$ Prolog application presented in Appendix D. Version control for both projects was managed with SourceSafe from One Tree Software.


[^0]:    1. A track is a path on a magnetic tape which stores a single channel of audio information.
[^1]:    1. In Audio Engineering, 'leakage' refers to microphones pick-up from sources other than that being recorded. Conversely, in signal processing the term refers to the spectral smearing introduced by windowing functions.
[^2]:    1. It should be noted that sufficiently slow attack times may lead to harmonic distortion in low-frequency tones.
[^3]:    1. Due to arrival time localisation, known as the Hass Effect, 'spatial' and level balance can differ significantly.
[^4]:    1. Advances in DSP have made other synthetic reverberation techniques obsolete.
[^5]:    1. Coincident microphone systems, such as the Calrec SoundField, are not strictly sound processing techniques.
[^6]:    1. For clarity, only one microphone input channel is shown.
    2. Of course, any linear audio potentiometer is still logarithmic in scale.
[^7]:    1. In most consoles, $n$ is a multiple of 4 .
[^8]:    1. REPLAY is the same as RECORD except that off-tape tracks come from the 'replay' head instead of the 'sync' head of an analogue multitrack machine, if connected.
[^9]:    1. Ultimation is something of a hybrid, allowing the motorised fader to be used purely as a level indicator, with the audio signal able to be switched silently and transparently between VCA and fader track as required.
[^10]:    1. Based on enhanced Z 80 processor technology.
[^11]:    1. As the dominant truncation takes place at one node.
[^12]:    1. Assuming, not unreasonably, both single-cycle MAC and multiply instructions.
[^13]:    1. Note that this is the sharing of controls and displays only - not the underlying processing resource.
[^14]:    1. Opponents of assignable mixing argue that one might want to touch any control instantly.
[^15]:    1. Superscalar architecture improve performance by exploiting additional instruction-level parallelism.
    2. COPAS-2 technology led to the development of the first commercial DMC, the Neve DSP1.
[^16]:    1. On-chip RAM can be accessed twice in each cycle, ensuring that DMA does not interfere with execution.
    2. In these TI DSPs, the second access can be used by some multi-operand instructions.
[^17]:    1. For a 4 G word address space per memory bank, the largest of all the DSPs
[^18]:    1. To which the TI TMS320C30 and TMS320C40 conform well.
[^19]:    1. Shadowing is also used by Analog Devices in the ADSP-21020 to achieve single-cycle context switching.
[^20]:    1. Products include the Peavey $D P M$ range of synthesizers and AMS/Neve Logic consoles.
[^21]:    1. Sony's stereo-format $8: 2 \mathrm{DMX}-\mathrm{E} 3000$ is based on the CXD serial-bus chip set discussed here.
[^22]:    1. Ariel's ISPW integrates two 1860 s with a Motorola DSP56001 on one NeXTcube expansion card
[^23]:    1. With no dedicated hardware, the T 8 xx has a multiply time of 13 instruction cycles, or 433 ns at 30 MHz !
[^24]:    1. In autonomous SIMD systems, individual processors can be masked during certain operations.
[^25]:    1. See Section 4.2.3, Processing Lag and Real-Time Operation.
[^26]:    1. Spanning-bus hypercubes should not be confused with the binary hypercube. The former, as the name suggests, uses buses connecting $w$ nodes, the latter uses point-to-point connections.
[^27]:    1. Due to the size of these tables, they have been repositioned as Appendix B.
[^28]:    1. The fall-through time of a FIFO is the time elapsing between the end of the first write to the FIFO and the time the first read may begin.
[^29]:    1. See the analysis of programmable DSP addressing modes in Section 4.3.4.
[^30]:    1. In that each node can support up to 17 DSP56001s with zero latency.
    2. IEEE International Conference on Acoustics Speech and Signal Processing.
[^31]:    1. As arcs are always assumed to be directed toward the exit node, they are not explicitly drawn.
[^32]:    1. For example, consider 2 processors and a taskforce of 3 independent tasks each requiring 2 instruction cycles.
[^33]:    1. Enough processors so that all tasks could be executed simultaneously if it were not for resource constraints.
[^34]:    1. In this context, the term 'even' means that if $m$ is a multiple of $n$, then each PE executes $m / n$ tasks. Otherwise, all but one executes $\lceil m / n\rceil$, with one PE executing the remaining tasks.
[^35]:    1. As found in architectures with a single common IPC channel, such as shared bus and simple ring topologies.
[^36]:    1. In any case, the curves intersect at $n=1$ and $n=m$.
[^37]:    1. An encoding scheme maps problem instances onto strings of symbols which describe them.
[^38]:    1. If $m=20$ and 10 million schedules evaluated per second, it would take almost 8000 years to check all 20 ! alternatives.
[^39]:    1. Analogous to a 'forest of trees'.
    2. Consequently, processors are never left idle if they can be assigned
[^40]:    1. A higher priority 'ready' task will pre-empt a partially processed lower priority task.
[^41]:    1. The reader should consult Appendix D for a complete listing of the DCS' Prolog source.
[^42]:    1. As a callee of $u t l_{-}{ }^{l b l} l_{-} f k(\ldots)$, utl_cnv_lvl(...) converts co-levels to pseudo-levels via $\omega_{c p}$
[^43]:    1. The prefix s_denotes the corresponding string as a $D C S$ internal symbol.
[^44]:    1. Normalised taskforce parallelism, represented in this thesis by $\pi$, must therefore be defined as $\frac{\omega_{s e q}}{n \omega_{c p}}$
[^45]:    1. In effect, an uninformed search based on dynamic programming.
[^46]:    1. NTL, the Netlist Translation Tool.
[^47]:    1. As $S(n)>S^{\prime}(n)$ for $1<n<m$, the ratio of $n$-fold parallel to sequential work must increase with $n$.
[^48]:    1. Of course, the same is true of NML IPC since only one concurrent inter-task message is required here.
[^49]:    1. Rather than a generic kernel, tailored code can reduce the cycle count for a 2 -input primitive by $41.67 \%$.
[^50]:    1. Irrespective of whether the $i$-input mix-bus is bus- or tree-configured.
[^51]:    1. Unfortunately, no lower bound can be derived for these dynamic or, indeed, other list-scheduling disciplines.
[^52]:    a. square-brackets indicate an optional argument.
    b. metric determines which topology metric should be returned to the worksheet.
    c. expansion determines whether the primary or secondary topology parameter should be expanded.

[^53]:    /* $============$
    TMPG headers
    /* $n============$

[^54]:    /* node expangion increment */
    alg_m_cllalgfnc( alg_f_frtnodinc ( $b, 1, n E x p C o d, ~ p d M e t v a l)) ; ~$ break;
    /* nodo oxpansion factor */
    alg_m_cllalgfnc( alg_f_frtnodfac ( $b, 1, \operatorname{nExpCod}$, pdMetVal)); break.

    ## \}ase MT_LNK INC:

    /* link oxpansion increment */
    alg_m_cllalgfnc( alg_f_frtlnkinc( b, $1, ~ n E x p C o d, ~ p d M e t V a l)) ; ~$ break;
    case MT_LNK_EAC:
    /* link oxpansion factor */
    alg_m_cllalgfnc( alg_f_frtlnkfac (b, 1, nExpCod, pdMetVall);
    break;

[^55]:    AlgErrCod
    AlgErrCod
    alg_f_nrsnhbmeh

[^56]:    /* node expansion increment */
    alg_m_cllalgfnci alg_f_ebhnodinc( $w, ~ d, ~ n e x p C o d, ~ p d M e t V a l l) ; ~$
     alg_m_cllalgfnc( alg_f_sbhnodfac (w, d, nExpCod, pdMetVal));
     alg_m_ellalgfncl alg_f_ebhinkinc( w, d, nexpCod, pdMetVal)); \} break;
    /* link expansion factor */
    alg_m_cllalgfnc( alg_f_sbhlnkfac( w, d, nExpCod, pdMetVal)); break;
    

[^57]:    Alatic
    algerrcod
    1

    ## Expnencod $\begin{gathered}\text { nExpCod } \\ \text { *pdMetVa }\end{gathered}$ <br> int int Expn

[^58]:    static
    IniErrcod
    ini_f_matertrcod

[^59]:     /* /* FUNCTION : xlo_f_betxlostx (...)
    /* NOTES : Set XLOPER string contents and type to xltypeStr /* INPUTS : ezstring - zoro-terminated atring */ $\begin{array}{llll}\text { /* INPUTS } & \text { : ezstring } & \text { - zero-terminated atring } \\ \text { /** } & \text { - } & \text { - optional argumenta } & * /\end{array}$

[^60]:    determ
    dyn_bgn_alg
    d_ipc_cod,
    d_lbl_cod,
    d_dyn_cod,

[^61]:    /* $==================$
    global predicates
    /* $==============$

[^62]:    

[^63]:    determ
    wri pro_lat

[^64]:    determ
    db-selector,

[^65]:    
    
    
    

[^66]:    include "dce_cpm.prv"

[^67]:    

[^68]:    | /** DESCRIPTION | : Non-destructive succeed if <TaskNamel> communicates |
    | :--- | :--- |
    | : directly with <TaskName2> |  |
    | /** |  |
    | /** PARAMETERS | : TaskName 1 |
    | /** TaskName2 |  |
    | /* CALLS | : SELF |

[^69]:    inf pretst (TaskList, Preclist, Proclist, (GranTest, FitTest, CritTest)):
    inf_grn_tet ( TaskList, ProcList, GranTest),
    inffit_tst( TaskList, ProcList, Fittost),
    inf_ctp_tst( PrecList, ProcList, CritTest).

[^70]:    

[^71]:    inf_sav_sch ( 0):-
    

[^72]:    $\dagger$. Reprinted in this thesis as Appendix F.

[^73]:    $\dagger$ Reprinted in this thesis as Appendix G.

