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A Case for Call Activity Detection for
ATM Voice Services in Project Parakeet

W.D. Blair

DSTO-TR-0606

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# A Case for Call Activity Detection for ATM Voice Services in Project Parakeet

*W.D. Blair*

**Communications Division**
**Electronics and Surveillance Research Laboratory**

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## ABSTRACT

The introduction of Asynchronous Transfer Mode (ATM) into the Australian Army tactical communication system will provide significant benefits through responsive network management and the integration of multiple communication applications. ATM integration into Parakeet is complicated as the current network employs military standard protocols which are quite different to civil standards. This report seeks to quantify capacity economies which can be achieved through the development of an adaptive approach to voice services on the ATM network by transmitting only active connections. An examination is also made to determine whether data communications protocols can access the released capacity. The study finds that savings are significant and the released capacity should add substantially to data communications capability.

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# A Case for Call Activity Detection for ATM Voice Services in Project Parakeet

## Executive Summary

The introduction of Asynchronous Transfer Mode (ATM) into the Australian Army tactical communication system will provide significant benefits through responsive network management and the integration of multiple communication applications. Integration of ATM with the current system (Parakeet) is complicated by the fact that Parakeet employs military standard protocols which are quite different to civil standards.

This report shows the significant bandwidth economies which can be gained if an adaptive approach to voice services on this ATM network by transmitting only active connections was developed. These bandwidth savings would then be released for use in general data communications providing substantial increases in communications capacity for automated command support systems.

Traffic analysis was carried out to determine the probability density function of the number of active calls in the busy hour. Three alternative designs for a tactical voice protocol are proposed here so that the bandwidth economy can be quantified. Given the proposed protocol, the bit rate required to carry the peak load and the average bit rate required in the busy hour can be determined and compared with the non-ATM approach. The results of this analysis show that, despite the imposition of additional protocol overheads from the ATM and other related protocols, substantial amounts of bandwidth would be released if the voice services over ATM were able to adapt to the number of channels in use. Even at peak usage the ATM approach expended only 432 kbps i.e. 16 % less capacity than the traditional Parakeet approach. Over the entirety of the busy hour the average ATM bit rate transmitted was 231 kbps , i.e. a further saving of 46 %.

Capacity released by the voice service responding to call and speech activity is intended to be used by Unspecified Bit Rate (UBR) traffic between automated command support systems. A brief examination is made that indicates that data communications protocols should be able to access the capacity released by call activity detection in this network. Nevertheless the wider issue of the operation of higher layer protocols over ATM UBR remains the subject of continuing study.

An example scenario was considered where the link between the two busiest headquarters was 1.024 Mbps. It was assumed that 256 kbps was set aside for video conferencing and other constant bit rate services (e.g. a superencrypted link). In such a scenario:

   if the traditional Parakeet approach was taken, voice services would use 512 kbps and data services (apart from circuit switched data connection via the voice services) would have access to a fixed rate of 256 kbps;

   if the voice services and all data were combined into an ATM stream, the Constant Bit Rate (CBR) services would consume 289 kbps (i.e. 256 bkps with ATM and related overheads) leaving 735 kbps for voice and data services:

if voice services were handled as a CBR ATM service they would consume 577 kbps leaving a fixed rate of 158 kbps for data (i.e. effective rate of 143 kbps after eliminating ATM overheads)

with call activity processing, voice services would consume during the predicted busy hour a maximum of 432 kbps and an average of 231 kbps. This would provide a data service with 303 kbps (during peaks) and 504 kbps on average (the effective rates after eliminating ATM overheads would be 274 kbps at worst and 447 kbps on average).

In summary, the value of ATM will only be achieved if the voice services in Parakeet employ a dynamic bandwidth approach.

# Authors

**W.D. Blair**
Communications Division

*William Blair is an Electronic Engineer in Network Integration Group of the Defence Science and Technology Organisation's (DSTO) Communications Division. He has been involved in research into the application of Asynchronous Transfer Mode (ATM) to military strategic and tactical communications since he joined DSTO from the Australian Army.*

# Contents

# 1. Introduction

The Australian Defence Force is currently procuring, under Project Parakeet, a new digital communications network for use by forces deployed to the battlefield. The system employs various transmission systems, especially satellite, to provide voice and data services between headquarters. It is based on a military standard time division multiplexed (TDM) system known as EUROCOM [1], which is similar in concept to the civilian integrated services digital network (ISDN). Voice systems are a fundamental element of a military tactical communications system, and comprise the central element of the Parakeet network; nevertheless, increasing data communications demands have pushed for increased data capacity in the network. As part of the next phase of Parakeet, a limited Asynchronous Transfer Mode (ATM) capability will be introduced to better integrate the voice and data uses of the network.

ATM promises to be the underlying protocol of the next generation of telecommunications systems. It exhibits some of the characteristics of circuit switched systems along with some of packet switched networks with the aim of providing a single network which can meet the needs of all users. The fundamental concept of ATM revolves around the transfer of information in small, fixed length packets called cells. The small size means that the time taken to fill the cell (cell fill latency) is kept relatively low. The fixed size means that hardware devices, once synchronised, can delineate and switch cells at very high speed. Because one to one (or one to many) circuits are established prior to transmission, as with circuit switching, the header does not need to uniquely identify end users; instead it identifies the connection. This means that the cell header can be considerably smaller than those required for packet networks so that while the cell is small, the header overheads do not become too burdensome. Like packet networks, cells from multiple sources can be multiplexed in a flexible fashion allowing users to obtain bandwidth on demand i.e. dynamic bandwidth allocation. To provide the link between user applications and the ATM cell layer, a series of alternative protocols forming the ATM Adaptation Layer (AAL) have been developed. Different AALs provide for the differing characteristics of the user application.

While there are definite management benefits from integrating voice services into the ATM network, the unsophisticated transmission of voice over ATM will incur bandwidth penalties via the addition of ATM headers and other AAL overheads to a Constant Bit Rate (CBR) digital voice stream. In the military tactical domain, where bandwidth is at a premium, this overhead makes the use of ATM less attractive compared to more traditional mechanisms such as TDM. Since a major strength of ATM is its ability to carry variable bit rate (VBR) traffic, the impact of the overheads associated with the ATM protocol might be reduced or eliminated if the voice traffic could adopt a VBR approach. There are two approaches:

- at the trunk level the capacity requirements can be varied to reflect the number of calls being connected at each moment (call activity); and

- at the call level the capacity requirement of each channel can vary according to the on-off activity in the conversation, i.e. speech activity (silence detection).

As discussed in a previous report [2], silence detection is the much harder approach and ought not be attempted in the first fielding of ATM in Parakeet. This report expands on the previous work, particularly in the context of call activity processing.

# 2. Aim

The aim of this report is to quantify the benefits of a voice service over Parakeet ATM that responds to call activity.

# 3. A Design for Voice Telephony over Military Tactical ATM

There are significant benefits in adopting civil standards in developing a tactical voice ATM system. Nevertheless, there will be some complicating issues arising out of the specific military environment and these are considered in this section. Such factors as latency and bandwidth efficiency need to be traded off in developing some alternative approaches and these are analysed further in this report.

## 3.1 The Need for Adaptation of Standards

In designing a protocol for the transmission of trunk voice channels over ATM in the military tactical domain, there are a number of factors which differentiate this environment from civil telephone systems:

- Whereas Pulse Code Modulation (PCM - the voice coding scheme for civilian systems) is an octet-oriented digital standard and many low bit rate schemes produce multi-octet frames (such as 9.6 kbps Code Excited Linear Prediction [CELP] which produces 36 octet frames), Continuously Variable Slope Delta modulation (CVSD - used in Parakeet) is bit oriented and the codec produces a stream of bits with each bit indicating a change to the waveform amplitude.

- Because the codec is operating at a low bit rate (16 kbps for Parakeet) protocol frame sizes which would be acceptable for civil systems (in terms of frame fill latency) may be unacceptable.

- While call activity or silence suppression mechanisms may not be implemented in the first introduction of voice over tactical ATM, the approach taken should not preclude its introduction as a future option.

- Because of bandwidth limitations, there should be minimal overheads.

## 3.2 Parakeet Architecture

Circuit switching technology is central to the Parakeet system. All services, including X.25 packet and military store and forward messaging, pass through the circuit switch. The network of circuit switches are interconnected with TDM trunks. While the capacity of these trunks can be set to one of a series of values from 256 kbps to 2.048 Mbps, trunks typically run at 512 kbps and the rate cannot be changed without disruption to ongoing calls. Meanwhile the user is provided with single channels with raw data capacity of 16 kbps. At the circuit switch it is possible to aggregate up to a maximum of six channels to produce a 96 kbps stream. 96 kbps is the upper limit and cannot be changed in a timely fashion.

The routing of calls from one user to another is via a flood search routing algorithm rather than by a deterministic mechanism. Connection requests are sent from the

originator's circuit switch on all connected links and are subsequently passed on by intermediate circuit switches until the circuit switch supporting the intended called party responds. The path of the response signal becomes the path for the circuit. This flood search routing algorithm will allow inter-connection of users without their knowledge of user physical locations.

Overall, while the Parakeet system is providing a quantum shift in capability compared to the system it is replacing, it does not provide the flexibility that modern data communications requires, especially in a bandwidth poor environment. As a consequence the decision was made to adopt ATM into the architecture.

In the initial introduction of ATM into Parakeet, only a limited number of nodes will be fitted with ATM. At an ATM fitted node the circuit switch system will continue to operate so that it can connect to the rest of the circuit switched network. Accordingly, the circuit switch will operate alongside the ATM. The most effective composite architecture is to use the ATM network to carry the circuit switch trunks and intelligently multiplex this trunk with newer applications.

As the Parakeet interswitch signalling is proprietary, it will be difficult to develop a solution which requires interpretation of the signalling. Coupled with the potential difficulties in coping with the Parakeet flood search routing algorithm, the use of the ATM system as a virtual transit switch is not considered a viable option. Since the ATM network does not route individual connections, calls from distant points in the network will continue to traverse a number of circuit switches under control of the flood search routing algorithm. At each entry to the ATM network, the transmission will suffer an ATM protocol data unit fill latency and this will accumulate with multiple links. Accordingly, approaches which minimise fill latency will be preferable.

While the interswitch signalling is not known, the trunk multiplexing waveform is defined in EUROCOM standards so individual channels could be demultiplexed in an interworking unit between the circuit switch and the ATM switch. If the indication pattern transmitted on idle channels can be identified then these channels can be suppressed to economise on bandwidth use.

The simplest mechanism to transport the voice traffic over the ATM would be to pass the entire 512 kbps trunk over an unstructured CBR virtual circuit using the appropriate civil standard AAL (AAL1). This is easy to implement, but offers no scope for any bandwidth economy measures to overcome the unavoidable ATM overheads. With a small increase in complexity, an approach could be taken where each voice circuit (plus the signalling channel) passes over an individual AAL1 virtual circuit. More sophisticated approaches would adopt or amend the multiplexed options described above.

## 3.3 Proposed Options

The overheads associated with the different protocol approaches varies as a function of frame size/frame fill latency and number of channels to be moved. Detailed comparisons are at Appendix A.

The EUROCOM trunk multiplex structure (for larger installations) is 512 kbps. This is made up of 30 channels of 16 kbps voice traffic, one channel of 16 kbps for interswitch common channel signalling and one 16 kbps synchronisation channel. This 512 kbps system is the baseline with which the various options are compared. For the

examinations I have assumed that the signalling channel will remain as a constant bit rate channel since the nature of its activity is not known.

Three AAL approaches are considered:

- **Dynamic Bandwidth Allocation using Circuit Emulation Service.** In this protocol there is a header comprising a bit-map identifying which TDM slots are being carried in the protocol data unit.

- **AAL1.** This option examines the concept of each voice call being passed on a unique AAL1 virtual circuit. The overhead of this protocol is simply the ATM cell header and one octet AAL1 sequence number. Partially filled cells are considered.

- **AAL for Composite Users (AAL-CU).** In this protocol there is a header which provides for fields such as call identity, user to user flags and header check sums.

Figure 1 provides a graphical comparison between the three options and describes the protocol data unit (PDU) structure (particularly the non-standard Dynamic CES approach) and how the structure is segmented into ATM cells for transmission over the ATM path. Each protocol has a header which results in overhead. In all cases larger voice channel frames will reduce the impact of this overhead but at the price of longer frame fill latency.

A key difference between the three AALs is the method used to identify the active TDM channels being passed via the ATM. Since the TDM trunk must be re-created at the end of the ATM path and active channels will be distributed throughout the available TDM channels, some method must be established in the various AALs to map transmitted data to its original TDM time slot. The Dynamic CES approach maintains the TDM time slot order, and uses the activity bit map in the PDU header to identify which channels appear in the PDU. The AAL1 approach uses pre-arranged virtual circuit identifiers to distinguish TDM channels. The AAL-CU has a channel identity field in the channel frame header which can be used.
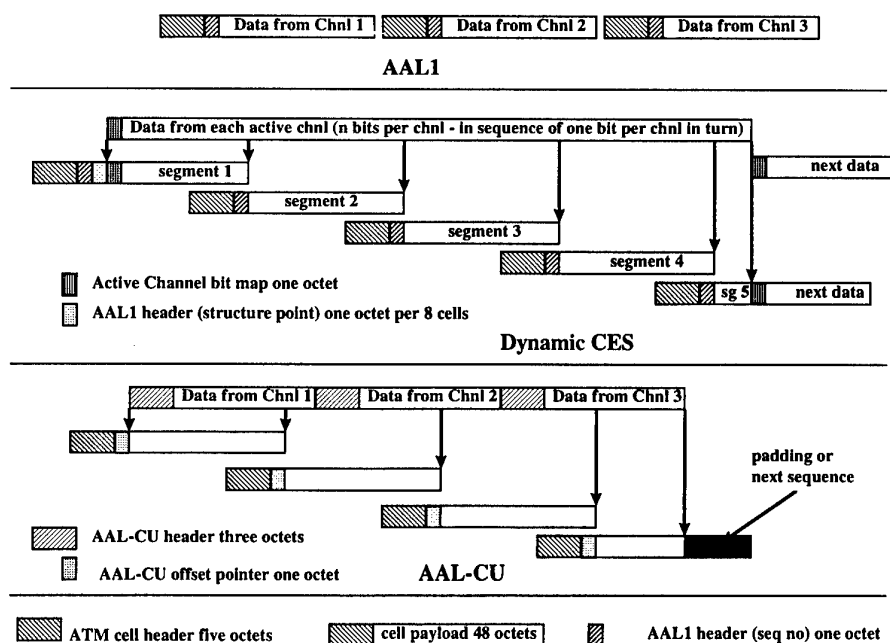


**Figure 1 - Protocol Data Units**

A solution employing an AAL1 virtual circuit per voice call is the simplest approach, but has unacceptable overheads unless complete cells are transmitted, thus incurring a 23.5 ms cell fill latency. This is borderline for acceptability, especially when calls need to transit several Parakeet switches, nevertheless it will be retained as one option.

Lower latencies lead to less efficient transmission in all protocol approaches. If a 5 ms latency is selected, then an approach employing dynamic bandwidth allocation using a modified Dynamic CES is the best solution and still provides quite an economic approach. This is used as the second option for analysis. Note that this approach, whilst based on the civil standard, would require custom development by the ATM switch vendor.

To use the civil AAL-CU standard, intended for low bandwidth real-time VBR services, a compromise latency must be selected. AAL-CU with 10 ms latency is used as the third option in the analysis.

Figure 2 shows graphically the bandwidth requirements of each option compared with TDM 512 kbps.
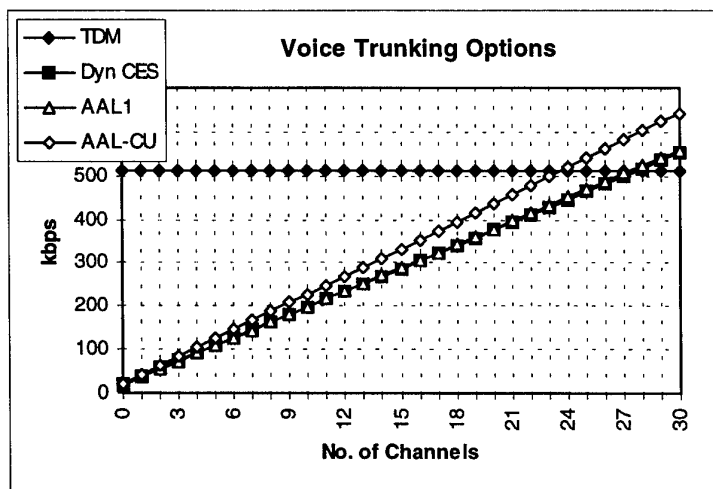


**Figure 2 - Parakeet Trunk Options**

An examination will be made later in this report on the number of channels that is likely to be required for an operational Parakeet network. Meanwhile, if a full 30 channel group is to be moved, ATM and protocol overheads will clearly mean the ATM options will require more bandwidth than the TDM solution. The bit rate required for various options is given in Table 1. (Note that the AAL1 option with one TDM channel per virtual circuit should not need to carry the synchronisation channel.)

**Table 1 - 30 Channel Bit Rate Requirements**

|  | AAL1 for Entire 512 kbps Trunk | Dynamic CES | AAL1 (one VC per TDM channel) | AAL-CU |
|---|---|---|---|---|
| Bit Rate Required | 577 kbps | 556 kbps | 559 kbps | 643 kbps |
| Overhead Compared to 512 kbps | 13% | 9% | 9% | 26% |

## 3.4 Summary

This section has considered the special factors arising from the military tactical environment, in particular recognising the Parakeet architecture. In trading off the two requirements of latency and bandwidth economy, it has proposed three protocols for the transmission of voice trunks in Parakeet. The Dynamic CES approach is a modification of the civil equivalent, but the other two are civil standards. All use more bandwidth than TDM for numbers of channels approaching the 30 traffic channels carried by the 512 kbps TDM trunk, but support VBR applications and use less bandwidth with fewer channels being transmitted. Later sections will determine the number of channels that would be expected to be moved simultaneously.

# 4. Capacity Management in Parakeet ATM

A key activity in managing the Parakeet ATM network, will be the allocation of capacity on the links between nodes. Central to such management in a large scale civil network is the concept of statistical multiplexing. This concept is explained in this section and subsequent sections will consider whether it will be significant in the Parakeet network. A concept is developed in this section for bandwidth management which provides a framework on which to base bandwidth savings estimates.

It is envisaged that there will be three major classes of users between which capacity must be shared for a particular link:

- **Voice Trunk.** This is a real-time service. While this traffic might be passed as a single virtual circuit or multiple virtual circuits, the composite should managed as a single entity.

- **Other Real-time Services.** These are services such as room based video conferencing and each should be managed individually.

- **Non-Real-time Services.** Typically these are tolerant of delay and delay variation, for instance computer interconnection for mail, file transfer etc.

## 4.1 Average Rate Management versus Peak Rate Management

In a large network, for which ATM was originally designed, a capacity management approach to VBR services was proposed based on statistical multiplexing. It is argued that if a large number of VBR sources are multiplexed the short term capacity peaks of each service would be decorrelated. Thus the total bandwidth demand would be less

variable than individual sources and tend to the sum of each source's average capacity requirement. Bandwidth could then be allocated to the composite based on this sum.

This approach is not recommended for Parakeet as statistical multiplexing becomes most effective with large numbers of sources and Parakeet will have few such VBR sources. Nevertheless, the statistical multiplexing effect will be considered in this study in respect of the combination of multiple VBR voice sources within the voice trunk.

Instead, an approach of peak rate management of real-time services is suggested for Parakeet. To meet quality of service requirements of real-time services, sufficient capacity must be allocated to meet the peak requirements. This ensures that cell loss and cell delay variation are minimised. Once all real-time services have been allocated their peak requirements the remaining capacity is in effect allocated for non-real-time services. In addition, capacity unused by real-time services operating at less than peak requirement, is available to non-real-time services.

## 4.2 An Approach at Looking at Bandwidth Savings

Figure 3 shows how capacity might be allocated between real-time services and non-real-time services. This also provides a model upon which bandwidth savings might be considered.

The peak requirement for voice services will be driven by call characteristics. Depending on the statistics of arrival rate and call holding time, the number of channels occupied (i.e. the number of simultaneous calls) will vary in accordance with a probability density function. From this function, the maximum number of calls can be determined.

Note that in calculating the capacity requirements of the system to forecast bandwidth savings, I have chosen not to dimension the voice service bandwidth to meet the absolute maximum requirement. Instead I have considered the 99.9% cumulative probability as the peak requirement. In effect this means that for 0.1% of time in the busy hour, the capacity considered in this analysis to be allocated to voice services would be insufficient and some calls may lose a short period of their transmission. In the worst instance, a call request at this moment may be disrupted. The 0.1% is consistent with, and will more than meet, the Parakeet performance specification which demands a grade of service for high priority voice users of 0.001 (i.e. 0.1%).

The difference between the peak bandwidth requirement and the bandwidth value that had originally been assigned to voice services (up to the maximum link capacity) is then a measure of the savings that can be allocated to non-real-time services with a reasonable degree of assurance that it will be available at all times. The difference between the peak bandwidth requirement and the actual bandwidth requirement at any moment in time will also be released to non-real-time services. However the amount released will vary in time driven by the changing call activity. Over a period, the amount released will be determined by the average bandwidth requirement of the voice service.
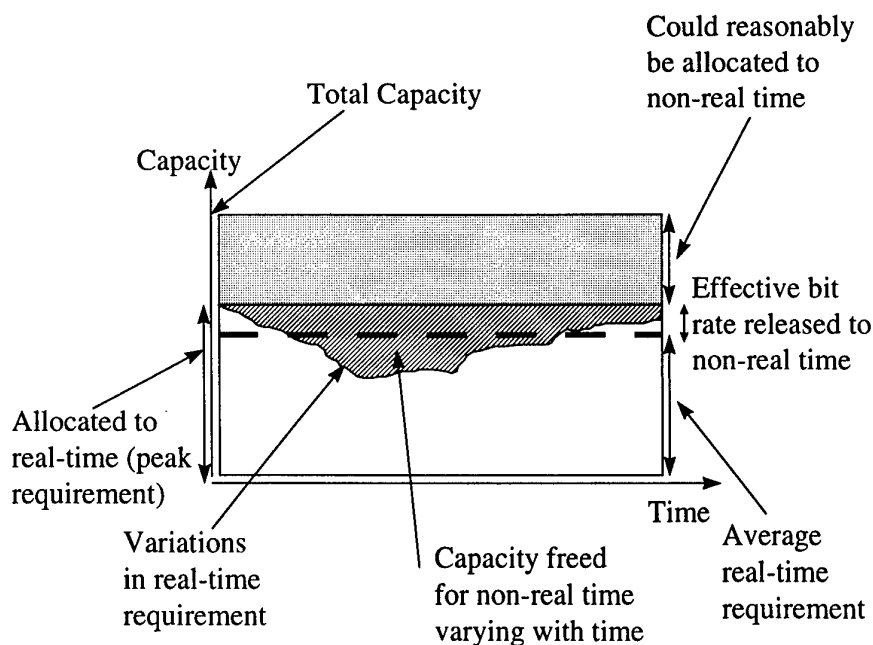
**Figure 3 - Capacity Allocation**

If one calculates the average capacity required for the voice services, one can determine the average capacity that would be unused and hence available for use to non-real-time services on an opportunity basis. This calculation is done by weighting the capacity requirements for each value of voice channels in use with the probability distribution of the number of channels being used.

## 4.3 Summary

It is argued here that statistical multiplexing, effective with large numbers, is not appropriate to Parakeet. Instead an approach using peak rate allocation is proposed. Capacity exceeding the peak rates requirements of real-time services can be assigned to non-real-time. The average requirement of the real-time services will reveal capacity which can released over a period to non-real-time services. The concept of peak usage and average usage will provide a framework for the description of bandwidth savings.

## 5. Bandwidth Savings from Parakeet Voice Call Activity

This section seeks to identify the bandwidth savings achievable through responding to connection activity. The section first carries out traffic analysis to determine the call activity characteristics of the busiest trunk in the Parakeet network. It then considers the bandwidth required to meet this traffic profile.

## 5.1 Determination of Voice Traffic Load

### 5.1.1 Method

In order to determine the bandwidth benefits from an ATM protocol responsive to call activity, the Parakeet traffic characteristics are required. Since Parakeet is producing a totally new capability for the Australian Defence Force, the usage of the network is not known with any great certainty. Traffic statistics were provided as part of the current Parakeet contract and these figures are taken as indicative of the operational network for the purpose of this study.

The Parakeet network topology is flexible and indeed dynamic, and would be responsive to the military operational situation at the time. For the purposes of the statistical analysis, traffic figures are provided for an indicative reference network layout shown in Appendix B. Since this network is equipped only with traditional circuit switches, and Parakeet is initially only equipping about six nodes with ATM, the author conducted discussions with the Parakeet project office to select the most appropriate nodes to be modelled with ATM.

The traffic load statistics from the Parakeet specification were processed to determine the busy hour call arrival rate for the busiest trunk link. This was then used in a conventional statistical model to determine the probability density function for the number of active channels in the busy hour.

### 5.1.2 Results

The detailed results were developed in two spreadsheets which are replicated in Appendix B. One spreadsheet analyses the call activity to determine busy hour call arrival rate and the second carries out statistical analysis (multifacility queuing system) to determine channel activity. The Parakeet reference network used for this analysis is shown in Figure 4. Results of each analysis are given here in Table 2.
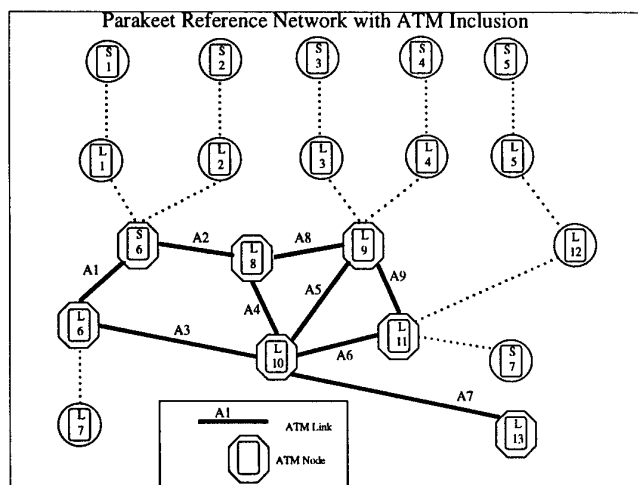


**Figure 4 - Parakeet Reference Network**

Table 2 - ATM Trunk Loading

| ATM Trunk Name | Parakeet Nodes Interconnected | Number of Calls initiated in the Busy Hour |
|---|---|---|
| A1 | L6/S6 | 69.375 |
| A2 | S6/L8 | 233.125 |
| A3 | L6/L10 | 130.625 |
| A4 | L8/L10 | 154.000 |
| A5 | L9/L10 | 157.750 |
| A6 | L10/L11 | 130.125 |
| A7 | L10/L13 | 85.000 |
| A8 | L8/L9 | 224.375 |
| A9 | L9/L11 | 130.625 |

Appendix C provides a table showing the probability density function of the call activity, this is shown graphically in Figure 5. Note the low probability of exceeding 23 active channels (99.9% i.e. less than 4 seconds in one hour).
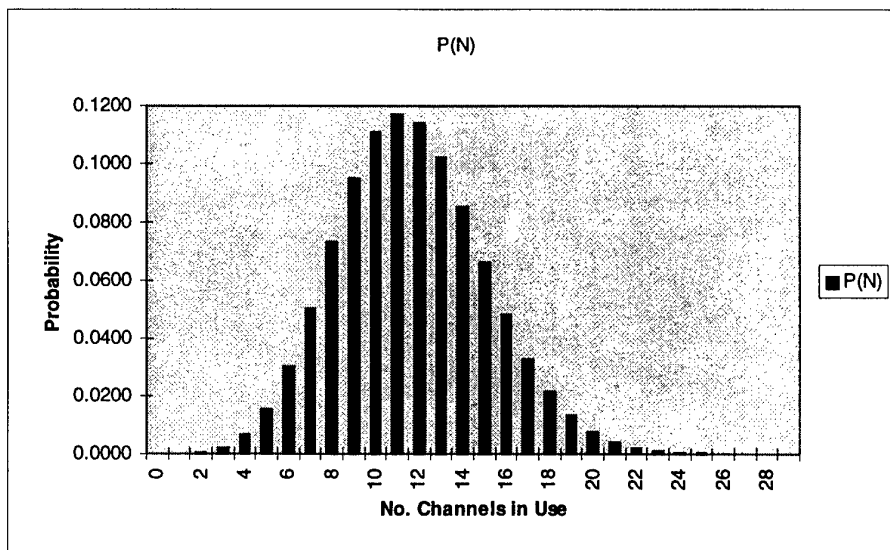


Figure 5 - Probability Density Function for Busy Hour Call Activity on the Busiest Trunk

## 5.2 Bandwidth Savings from Dynamic Bandwidth Use in Response to Call Activity

Two calculations have been carried out (detailed table of results at Appendix C):

- **Assuming 23 channels in use.** This is equivalent to the worst case, i.e. sufficient channels to cater for 99.9% situation on the busiest trunk in the busy hour. The bit

rate requirements from this situation should be reserved for voice use in a peak rate management approach. Any excess transmission capacity above this peak rate can reasonably be assigned to high priority non-voice uses.

- **Over the entire probability range of Channels in Use.** This determines the weighted average bit rate by considering the bit rate required for each channel usage situation and weighting that figure by the probability of that number of channels being required. The resultant bandwidth requirement is then assessed against:

  - **The ATM Trunking Requirements for 23 Channels.** This provides a measure of the bandwidth that would be released in an ATM system to low priority uses, and

  - **512 kbps.** This provides a measure of the overall savings compared with TDM that would be achieved through the adoption of that ATM trunking approach).

**Table 4 - Bandwidth Savings from Call Activity**

**Bandwidth Savings Compared to 512 kbps when 23 Calls are Active**

|  | Dyn CES | AAL1 | AAL-CU |
|---|---|---|---|
| B/W Reqt for 23 Chan | 432.2 kbps | 433.0 kbps | 498.0 kbps |
| Saving | 16% | 15% | 3% |

**Bandwidth Savings Through Call Activity Detection**

|  | Dyn CES | AAL1 | AAL-CU |
|---|---|---|---|
| Weighted Average B/W Reqt | 231.3 kbps | 228.3 kbps | 262.6 kbps |

**%age saving on 23 channel requirement**

| Dyn CES | AAL1 | AAL-CU |
|---|---|---|
| 46% | 47% | 47% |

**%age saving on 512 kbps**

| Dyn CES | AAL1 | AAL-CU |
|---|---|---|
| 55% | 55% | 49% |

## 5.3 Summary

These figures indicate significant savings, particularly in the case of Dynamic CES and AAL1. In these cases the bandwidth requirement for the predicted busy hour, busiest link channel occupancy of 23 channels is clearly less than the 30 channel TDM requirements. By responding to call activity, even this peak requirement has more than overcome the ATM and AAL overheads and provides savings. Furthermore,

substantial savings are available over the course of the busy hour to the tune of about half of the capacity that would otherwise have been reserved exclusively for voice services.

# 6. Ability to Make Use of Released Capacity

In the concept of bandwidth management proposed in this report, capacity for the voice and other real-time services is reserved based on a peak rate. This will ensure that cell loss and cell delay variation are minimised for real-time services. Capacity released by the voice service responding to call activity is intended to be used by Unspecified Bit Rate (UBR) traffic. It is not immediately apparent whether higher layer data communications protocols have the ability to respond adequately to make use of the changing capacity. Variations in channel usage which derive the variations in spare capacity will cause variations in UBR cell transmissions. Higher layer data communications protocols will perceive these variations as variations in the round trip time (RTT) of acknowledgments of transmitted packets. When capacity released to UBR is high then cells are released quickly with a consequent reduction in the time taken for packets to reach their destination and acknowledgments to be returned.

In examining the ability of higher layer protocols to take advantage of the capacity released by VBR voice services two questions must then be asked: what is the nature of the variation in capacity - as perceived by the higher layer protocol; and how will this impact on the higher layer flow control strategy. For the purposes of this study, the higher layer protocol considered is the internet Transmission Control Protocol (TCP).

## 6.1 Data Capacity Available

For the purposes of this analysis, a small simulation was developed depicting the busy hour on the busiest trunk. Call activity was generated in the model using the traffic statistics determined earlier. A record was then made of the number of active channels in each time slot (each of 1 second) of the simulation. An extract of that record is shown in Figure 6 to show the time scales of the capacity changes. It was chosen for
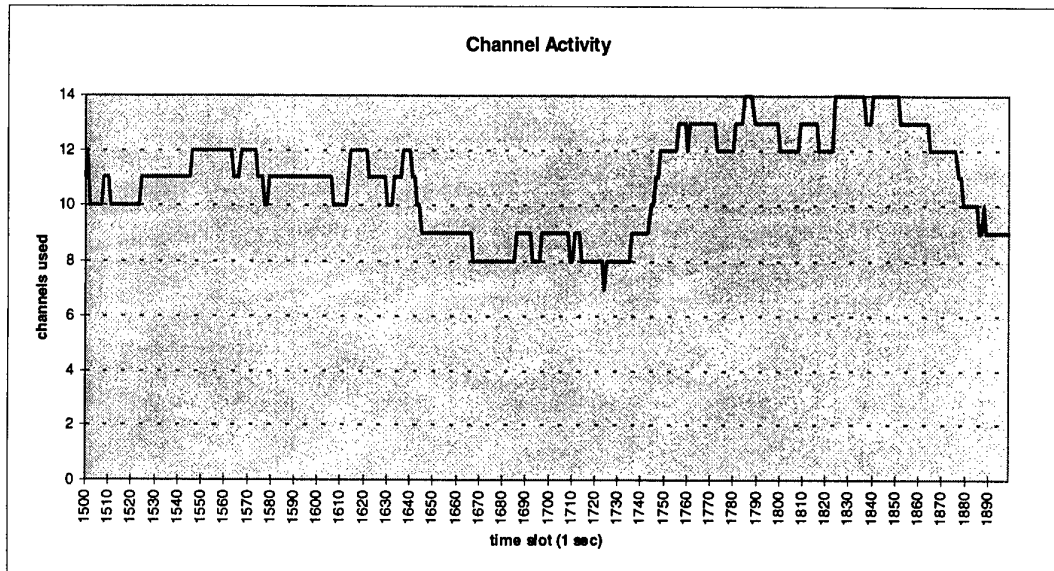


**Figure 6 - Time Series Showing Number of Channels Active**

display because of the large variation in channel usage and to see the impact of this on TCP.

As the number of active channels varies, the bandwidth consumed by the voice services changes (with the precise demand dependent on the AAL employed). To consider how this variation impacts on the capacity available to data communications one must know the total capacity available for voice and data. Parakeet satellite terminals are limited to a 2.048 Mbps throughput, but typically this capacity would be shared between two links. For the purposes of this analysis, the 1.024 Mbps for a link is assumed to allocated with 256 kbps set aside for video conferencing and other CBR services (consuming almost 289 kbps with ATM/AAL1 overheads) with the remaining 735 kbps shared between voice services and data.

The capacity available to data communications can now be examined. This was calculated for the simulation using the Dynamic CES option. A short portion of the results showing the corresponding period as Figure 6 is shown in Figure 7. Also shown is the capacity that would have been available to data if the voice service did not respond to call varying activity thus consuming 577 kbps (512 kbps with ATM/AAL1 overheads).
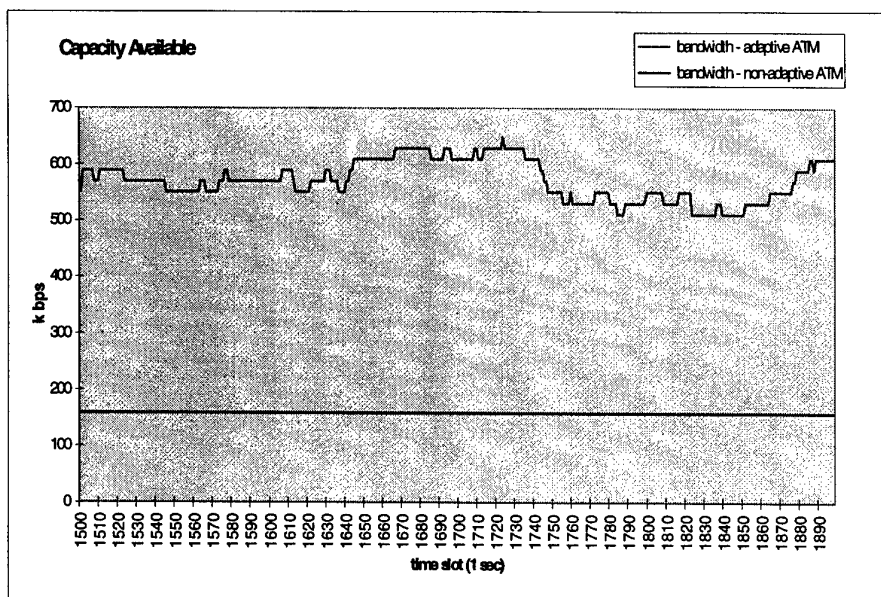


Figure 7 - Time Series Showing Data Capacity Available

## 6.2 Impact on TCP

TCP is a fundamental protocol within the Internet and intrinsic to the protocol is a mechanism to cope with variations in RTT caused by variations in congestion. The throughput of TCP is a complex interaction between the TCP acknowledgment window (the number of transmitted packets awaiting acknowledgment) and transmission timeout interval. Of particular concern is the transmission timeout as, if a packet is not acknowledged before timing out, TCP will assume congestion, reduce its

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transmission rate and retransmit all packets in the window. RTT variation can impact on throughput as RTT is used by TCP to set timeout intervals for waiting for acknowledgments. Short term variations in RTT are handled through two aspects of typical TCP implementations as conceived by Van Jacobson and described in Tanenbaum [3] Chapter 6:

- The timeout is based on an exponentially smoothed figure for estimated RTT. The estimated RTT is revised with the arrival of most packets and is the weighted average of the previous estimate and the measured RTT for that packet. The previous estimate has the greater weight, typically 7/8.

- The timeout is typically set to the estimated RTT plus (typically) four times the mean deviation in RTT.

In an ATM system, non-real-time services attract a lower priority in the scheduling of cells onto the transmission link. To reduce the possibility of cell loss, such services are handled through relatively long buffers. In the case of the DSTO research ATM network, these buffers are of the order of 1000 cells (about 380,000 bits of user data). As link capacity available to the non-real-time services waxes and wanes, the length of queued data in the buffer will consequently and inversely wane and wax. Clearly as the queue enlarges, the transit time for the TCP packet to traverse the network will increase and the time taken for acknowledgments to be received at the sending process will increase. Increasing delay in receiving acknowledgments has a secondary impact by slowing traffic offered by TCP as new traffic cannot be offered until outstanding packets are acknowledged.

The most catastrophic thing that can happen to a TCP stream is for a packet acknowledgment to timeout. This can occur either through the RTT exceeding the timeout limit or if the packet is dropped, either in whole by an IP router or in part by the overflow of an ATM link buffer. If the TCP acknowledgment times out, TCP drastically reduces the amount it is prepared to send in each burst. The rate at which this amount is allowed to increase is initially quite fast, but once a threshold is reached (determined by what had been happening on the network in the immediate past) the rate of increase becomes quite slow.

The impact of the varying data capacity was examined in the simulation. A simple model was developed with a single TCP source and terrestrial transmission path. This was chosen so that the impact of the varying data capacity was not hidden by other factors such as interaction between multiple TCP processes or the impact of satellite delay on TCP. (These are covered in the general literature with some interesting analysis in Goyal [4]). An examination of the performance of this simple model shows:

- The ATM buffers are quite large compared to the variation in available capacity (buffers are of the order of 380 kbits while the standard deviation of the data capacity figure is less than 64 kbps). While such buffers are insufficient to cope if TCP was transmitting at fixed rate equal to the average available capacity, they have shown to be sufficient to absorb the mismatches between the adaptive TCP rate and the available bandwidth. ATM buffers do not suffer overflow in the model.

- In this model, when the TCP stream starts it has a poor estimate of RTT. Consequently the initial transmissions suffer a number of timeout failures. Very soon the estimate, and the RTT deviation, become a more accurate representation of the network performance and timeouts no longer occur.

- After the timeouts are resolved, the transmission burst size slowly increases until the maximum is gained, and remains at that size for the remainder of the simulation. This is the logical outcome from the situation where there are no further timeouts caused by excessive acknowledgment delay or loss of transmission through buffer overflow.

- Buffer queue length and available bit rate then determine the RTT and hence time before TCP can send more traffic. This effectively modulates the transmission of TCP packets into the network. Figure 8 depicts the TCP goodput (i.e. the size of successful TCP transmissions commenced in each time slot) for a segment of the simulation.

- Average available bit rate for this simulation is 581 kbps. The TCP stream averages 480 kbps.
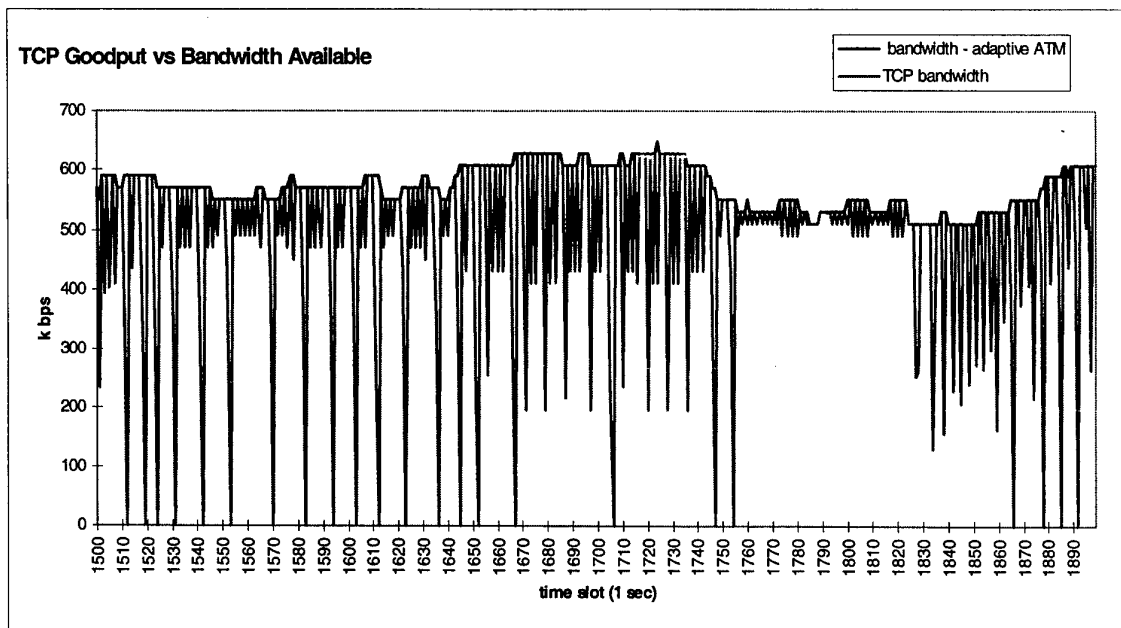


Figure 8 - Time Series Showing TCP Goodput

The mechanisms conceived by Van Jacobsen would therefore appear to be sufficient to cope with the variability in capacity for the system considered in this report. Nevertheless the wider issue of TCP over ATM UBR, especially over satellite and over congested networks, remains an open question and a subject of continuing study see Goyal [4].

## 6.3 Summary

This section has considered whether the capacity released by call activity detection can be effectively used by higher layer data protocols, specifically TCP. With the in-built mechanisms to cope with short term variability in network capacity available in TCP,

there is reason to have some confidence in the ability of TCP to use much of the released capacity.

# 7. Conclusions

The introduction of ATM into traditional TDM communication systems provides significant benefits in the areas of responsive network management and the integration of multiple communication applications. The penalty which comes with this benefit is the additional overheads intrinsic to ATM and associated protocols. This report seeks to quantify capacity economies which can be achieved through the development of a VBR approach to the passage of voice services through the ATM network transmitting only those channels between circuit switches which have active connections.

Parakeet has been designed for use in a tactical (mobile) military environment. It employs military standard protocols which, while similar in function to civil standards, are quite different in implementation. Accordingly, in order to quantify the bandwidth economy measures, three alternative designs have been proposed as a tactical voice protocol. These are either direct civil implementations or based on civil standard approaches.

To compare the bandwidth requirements of a VBR approach with that using a fixed allocation two measures of bandwidth savings are defined:

- The first determines the peak bandwidth requirement of the VBR approach and compares this to the fixed allocation. This saving ought never be intruded upon and can be reasonably allocated to other services.

- The second calculates the average bandwidth requirement of the VBR approach and compares this to the peak allocation. This capacity waxes and wanes over time and hence can only be allocated to non-real-time uses.

Traffic statistics from the Project Parakeet contract were analysed to determine busy hour traffic arrival rate and call holding times for the busiest trunk. Queuing theory (multifacility queuing system) analysis was carried out and this showed that peak channel occupancy (99.9% level) was only 23 out of the 30 channels available on that link. This provides considerable opportunity for savings. With the preferred voice trunking options:

- capacity required for peak activity is around 15% lower than the traditional TDM system

- average capacity in the busy hour is around 46% lower than the requirement for the peak load.

Capacity released by the voice service responding to call activity is intended to be used by Unspecified Bit Rate (UBR) traffic. Variations in channel usage which derive the variations in spare capacity will cause variations in UBR throughput. The higher layer data communications protocols will perceive these variations through variations in the round trip time (RTT) of acknowledgments of transmitted packets. It is not immediately apparent whether higher layer data communications protocols (particularly TCP) have the ability to respond adequately to such changing capacity. An examination of the simulation developed for this report suggests that the capacity

is useable given the algorithms inbuilt with TCP to cope with variations in RTT. Moreover, while there is still work being done in assessing the ability to use all of the capacity released by these techniques, it is important not to lose sight of the significant capacity released when considering the peak requirements alone.

# 8. References

[1]     "Tactical Communications Systems Basic Parameters", EUROCOM D/1.

[2]     W.D. Blair, "Project Parakeet - Integration of Voice Services with a Tactical Asynchronous Transfer Mode Network" DSTO Report DSTO-TR-0541, 1997.

[3]     A. S. Tanenbaum, *Computer Networks*, Third Edition, New Jersey: Prentice Hall International.

[4]     R. Goyal, R. Jain, S. Kalyanaraman, S. Fahmy, B. Vanalore, X. Cai, S-C Kim and S. Kota. "Guaranteed Rate for Improving TCP Performance on UBR+ over Terrestrial and Satellite Networks" ATM Forum Document: ATM_Forum/97-0424, Apr 1997.

[5]     ATM Forum Technical Committee Baseline Document atm96-0136R, "Specification of Dynamic Bandwidth Utilization in 64 Kbps Time Slot Trunking over ATM Using CES." 14 November 1996.

[6]     ATM Forum Technical Committee AF-SAA-0032.000, "Circuit Emulation Service Interoperability Specification", September 1995. Subject to replacement by ATM Forum Technical Committee AF-VTOA-0078.000, "Circuit Emulation Service Interoperability Specification Version Two, Letter Ballot", November 1996.

[7]     Draft new ITU-T Recommendation I363.2, B-ISDN ATM Adaptation Layer Type 2 Specification", Madrid, November 1996.

[8]     H. J. R. Dutton and P. Lenhard, *Asynchronous Transfer Mode (ATM) Technical Overview*, New Jersey: Prentice Hall PTR.

[9]     E. Turban and J. R. Meredith, *Fundamentals of Management Science.* Burr Ridge, Illinois: Irwin.

# 9. Abbreviations

| | |
|---|---|
| AAL | ATM adaptation layer |
| ATM | Asynchronous Transfer Mode |
| CBR | constant bit rate |
| CVSD | continuously variable slope delta modulation |
| ISDN | integrated services digital network |
| ITU | International Telecommunications Union |
| RTT | round trip time |
| TCP | transmission control protocol |
| UBR | unspecified bit rate |
| VBR | variable bit rate -rt for real-time and nrt for non-real-time applications |

# Appendix A: Parakeet Voice Trunk Options

## 1.1 Introduction

Figure A-1 shows diagrammatically the three options developed for this study.
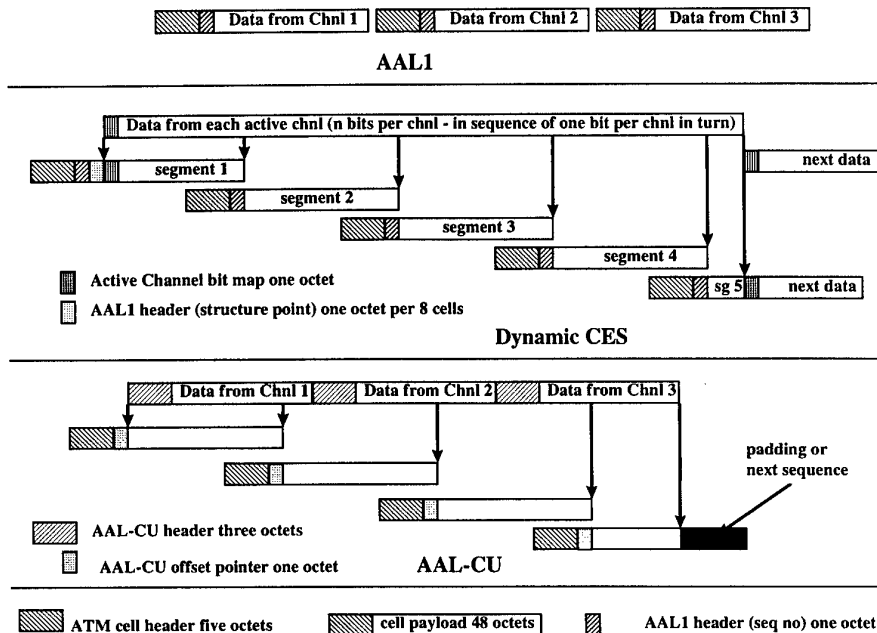


**Figure A-1: Three Trunk Options for Parakeet**

A key difference between the three AALs is the method used to identify the active TDM channels being passed via the ATM. Since the TDM trunk must be re-created at the end of the ATM path and active channels will be distributed throughout the available channels, some method must be established in the various AALs to map transmitted data to its original TDM time slot.

### 1.1.1 AAL1

This option passes each voice call plus the signalling channel on a unique AAL1 virtual circuit. Prearranged virtual channel identifiers can be assigned to map channels on the TDM trunk to ATM virtual circuits. The ATM/AAL overhead of this protocol is simply the ATM cell header and one octet AAL1 sequence number. Since an integral number of cells must be transmitted, overheads are minimised for a frame size equivalent to the AAL1 payload (47 octets = 376 bits) or an integral multiple of payloads (this would eliminate wasteful bit padding). The number of 53 octets cells per PDU is thus given by frame size/376 rounded up to the next integer.

The rate at which PDUs are produced by each channel is the individual channel rate divided by the frame size. Since the voice channel bit rate is 16 kbps, the PDU production rate is 16/frame size kPDU per second per channel. The PDU fill latency encountered by each channel is the inverse of the figure.

The overall bit rate of this AAL is thus given by the following:
bits per cell * cells per PDU * PDU production rate per channel * number of channels
(53*8) * (frame size/376 rounded up) * (16/frame size) * number of channels kbps
(Note the number of channels is the number of traffic channels plus one channel for signalling.)

## 1.1.2 Dynamic CES

This option passes the entire trunk (all voice calls plus the signalling channel) on a single AAL1 virtual circuit. The PDU has a four octet header representing a bit-map to identify which TDM slots are being carried in the body of the protocol data unit. The ATM/AAL overhead of this protocol is the ATM cell header, one octet AAL1 sequence number per cell, one octet AAL1 structure pointer (to point to the start of a PDU) per eight cells and the bit map octets per PDU. The structure pointer is required as the PDUs are not synchronised to always start at the beginning of a cell - thus there is no need for padding.

This AAL is a derivative of the civil equivalent [5]. Whereas the civil system transmits octets of voice channel data (because PCM is octet based), the proposed tactical design transmits individual channel data as bits (as done in the EUROCOM TDM frame [6]). A frame size (n) for each channel is defined as the number of bits being carried in the PDU for a channel. The Dynamic CES structure will then be four octets of header plus n bits per channel (i.e. the number of active channels plus one channel of signalling). The data portion of the structure is one bit per channel in sequence through the channels being carried. The AAL1 structure pointer must point to the bit map and this must appear on an octet boundary; this is most easily arranged by having each channel frame to be an even number of octets.

Because of the structure pointer, the number of octets per cell is on average 47.875. The average number of 53 octet cells produced by a Dynamic CES PDU is:

(4 + (channel frame size in octets) * (number of channels) )/47.875

The rate at which PDUs are produced is the individual channel rate divided by the frame size (in bits) per channel. Since the voice channel bit rate is 16 kbps, the PDU production rate is 16/frame size (in bits) kPDU per second. The PDU fill latency encountered by each channel is the inverse of the figure.

The overall bit rate of this AAL is thus given by the following:

bits per cell * cells per PDU * PDU production rate per channel * number of channels
(53*8) * ((4 + (frame size in octets) * (number of channels) )/47.875) * (16/frame size in bits) kbps
(Note the number of channels is the number of traffic channels plus one channel for signalling.)

## 1.1.3 AAL-CU

The option uses the civil standard [7] and passes the entire trunk (all voice calls plus the signalling channel) on a single ATM virtual circuit. Associated with each channel frame is a three octet header which carries, amongst others, a channel identity field which can be used to identify the matching TDM slot. The ATM/AAL overhead of this protocol is the ATM cell header. one octet AAL-CU structure offset pointer (to point to

the start of a channel frame) per cell and the AAL-CU header per channel frame. The offset pointer is required as the channel frames are not synchronised to always start at the beginning of a cell. Only if the next channel frame is not formed within an acceptable delay of the current channel the last cell will be transmitted with padding. For the purposes of this study, I have assumed that there will be no instances of padding being required.

Because of the offset pointer, the number of octets per cell is 47. The number of 53 octet cells produced by AAL-CU is:

(3 + (channel frame size in octets) * (number of channels) )/47

The rate at which channel frames are produced is the individual channel rate divided by the frame size per channel. Since the voice channel bit rate is 16 kbps, the PDU production rate is 16/frame size (in bits) k channel frames per second. The frame fill latency encountered by each channel is the inverse of the figure.

The overall bit rate of this AAL is thus given by the following:

bits per cell * cells per PDU * PDU production rate per channel * number of channels
(53*8) * ((3 + (frame size in octets) * (number of channels) )/47) * (16/frame size in bits) kbps
(Note the number of channels is the number of traffic channels plus one channel for signalling.)

## 1.2 Protocol Performance Comparison

When the frame for each channel is of the order of an ATM cell payload (parameters are given in Table A-1), all techniques are similar in performance (in order: AAL1, dynamic CES then AAL-CU). As seen in Figure A-2, until the trunk load exceeds 25 (for AAL-CU), 27 (for dynamic CES and AAL1) the ATM techniques are more bandwidth efficient than TDM in spite of the overheads. The frame fill latency is such that a typical worst case of six trunk hops, assuming no satellite links, would remain below the 150 ms latency quoted by Dutton [8 page 8-12] as the maximum suggested for voice conversations.

## Table A-1 - Voice Protocol Parameters .-.Case One

### Parameters (in bits)

| | |
|---|---|
| 32 | D-CES bitmap |
| 352 | D-CES frame per chan |
| 22 | frame latency (ms) |
| | |
| 376 | AAL1 frame |
| 23.5 | frame latency (ms) |
| | |
| 24 | AAL-CU header |
| 352 | AAL-CU frame |
| 22 | frame latency (ms) |
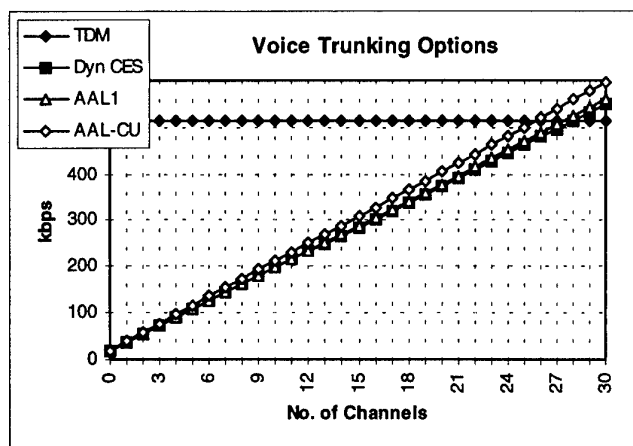


**Figure A-2 - Bandwidth Requirements for Each Option - Case One**

## Table A-2 - Voice Protocol Parameters - Case Two

### Parameters (in bits)

| | |
|---|---|
| 32 | D-CES bitmap |
| 80 | D-CES frame per chan |
| 5 | frame latency (ms) |
| | |
| 80 | AAL1 frame |
| 5 | frame latency (ms) |
| | |
| 24 | AAL-CU header |
| 80 | AAL-CU frame |
| 5 | frame latency (ms) |
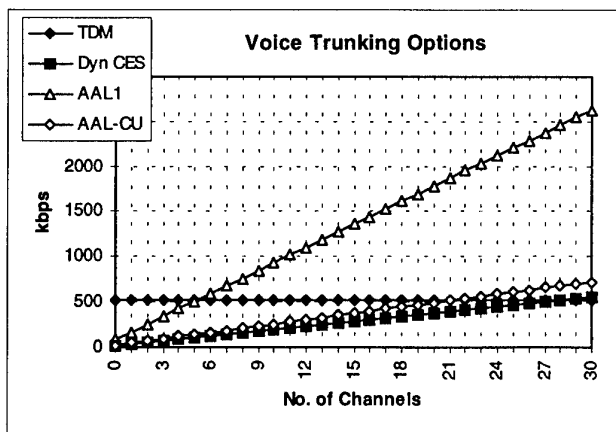


**Figure A-3- Bandwidth Requirements for Each Option - Case Two**

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For smaller frames per call, such as shown in Table A-2 and Figure A-3, AAL1 becomes unacceptably inefficient. For instance at a frame latency of 5 ms AAL1 exceeds the baseline figure when more than 5 channels are active. Dynamic CES is best performing being better performing than the baseline until more than 27 channels are active while the AAL-CU exceeds the baseline after 21 channels.

If a compromise frame fill latency of 10 ms is chosen (Table A-3 and Figure A-4) AAL1 is significantly improved but still unacceptably inefficient exceeding the baseline figure at more than 11 channels active. Dynamic CES remains best performing having lower bit rate than the baseline until more than 27 channels are active while the AAL-CU now exceeds the baseline at more than 23 channels.

**Table A-3 - Voice Protocol Parameters - Case Three**

| Parameters (in bits) | |
|---|---|
| 32 | D-CES bitmap |
| 160 | D-CES frame per chan |
| 10 | frame latency (ms) |
| | |
| 160 | AAL1 frame |
| 10 | frame latency (ms) |
| | |
| 24 | AAL-CU header |
| 160 | AAL-CU frame |
| 10 | frame latency (ms) |
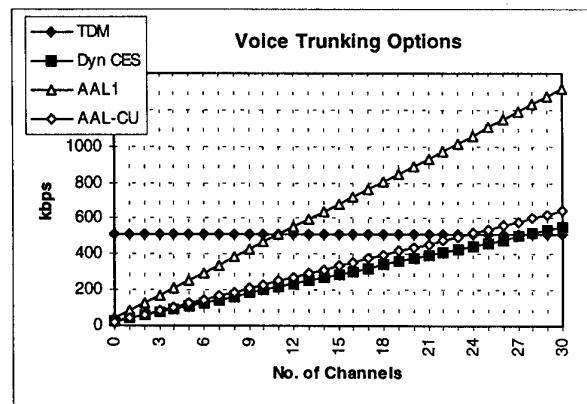


**Figure A-4- Bandwidth Requirements for Each Option - Case Three**

## 1.3 Protocol Parameters Selected for Study

The following parameters were selected as a basis for further examination in this study:

- Dynamic CES: active channel bit map of 32 bits, 80 bits per channel data frame
- AAL1: 376 bits per channel data frame
- AAL-CU: standard AAL-CU header of 24 bits, 160 bits per channel data frame

The resultant kbps requirement for the range of active channels is given in table A-4.

**Table A-4 - Voice Protocol Bandwidth Requirements (kbps) - Options for Further Examination**

| Chnls | Dyn CES | AAL1 | AAL-CU | Chnls | Dyn CES | AAL1 | AAL-CU |
|---|---|---|---|---|---|---|---|
| 0 | 24.8 | 18.0 | 20.7 | 16 | 308.2 | 306.7 | 352.7 |
| 1 | 42.5 | 36.1 | 41.5 | 17 | 325.9 | 324.8 | 373.5 |
| 2 | 60.2 | 54.1 | 62.2 | 18 | 343.6 | 342.8 | 394.2 |
| 3 | 77.9 | 72.2 | 83.0 | 19 | 361.3 | 360.9 | 415.0 |
| 4 | 95.6 | 90.2 | 103.7 | 20 | 379.1 | 378.9 | 435.7 |
| 5 | 113.4 | 108.3 | 124.5 | 21 | 396.8 | 396.9 | 456.5 |
| 6 | 131.1 | 126.3 | 145.2 | 22 | 414.5 | 415.0 | 477.2 |
| 7 | 148.8 | 144.3 | 166.0 | 23 | 432.2 | 433.0 | 498.0 |
| 8 | 166.5 | 162.4 | 186.7 | 24 | 449.9 | 451.1 | 518.7 |
| 9 | 184.2 | 180.4 | 207.5 | 25 | 467.6 | 469.1 | 539.5 |
| 10 | 201.9 | 198.5 | 228.2 | 26 | 485.3 | 487.1 | 560.2 |
| 11 | 219.6 | 216.5 | 249.0 | 27 | 503.0 | 505.2 | 581.0 |
| 12 | 237.4 | 234.6 | 269.7 | 28 | 520.8 | 523.2 | 601.7 |
| 13 | 255.1 | 252.6 | 290.5 | 29 | 538.5 | 541.3 | 622.5 |
| 14 | 272.8 | 270.6 | 311.2 | 30 | 556.2 | 559.3 | 643.2 |
| 15 | 290.5 | 288.7 | 332.0 | | | | |

# Appendix B: Parakeet Trunk Loading Calculations

Figure B-1 provides the Parakeet Reference Network used for the traffic model. Note that only six nodes are ATM equipped and hence subject to the analysis.

The traffic load statistics from the Parakeet Specification provides figures in terms of call source and destination and are not expressed in terms of individual trunk loadings. The figures were provided as daily total call initiations and had to be converted to busy hour figures using guidance from the Parakeet Specification (busy hour was stated to be six times the average per hour rate).

The Parakeet network employs flood search routing, this will result in the call being connected via the shortest open path. For the purposes of this analysis, each source-destination node pair was examined and this traffic was allocated to a series of concatenated trunks following the shortest path. If there were equidistant paths, the traffic was evenly divided between the two alternatives.

Once traffic between the nodes was allocated to various trunks, the load on each trunk can be determined. For the purposes of this analysis only trunks between ATM nodes were subject to further consideration. Only the highest loading trunk was examined in the next portion of the model.

Traditional queuing statistics (multifacility queuing system from Turban and Meredith [9] p644) was then used to calculate the probability density function of the number of calls at each moment.
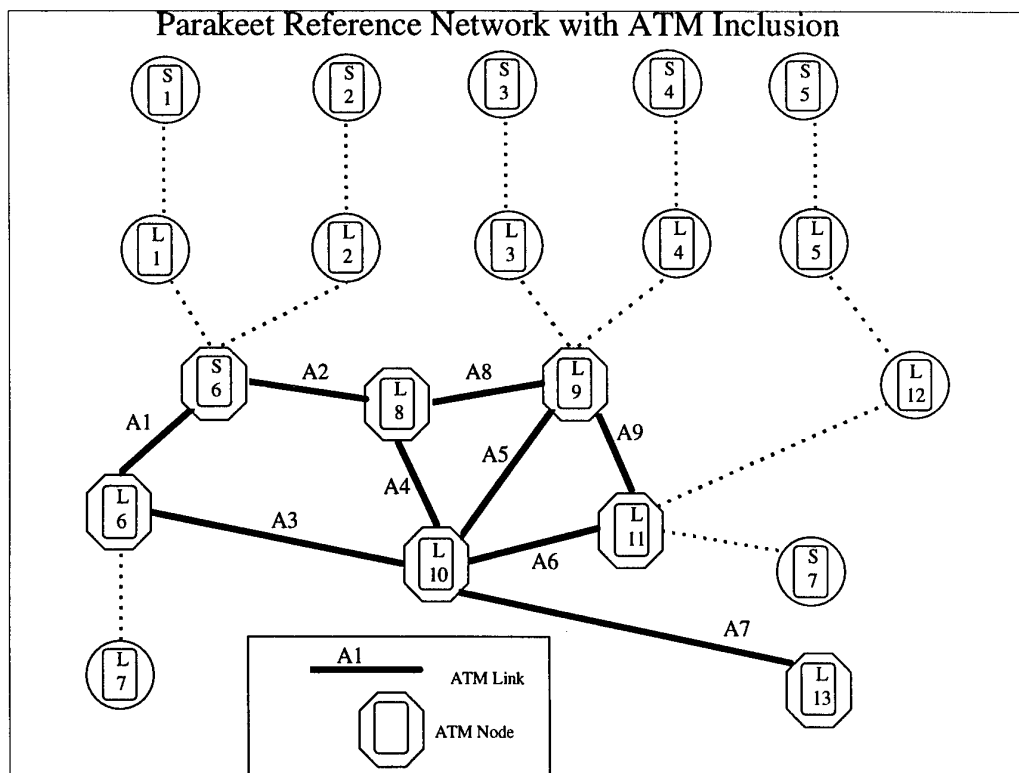


Figure B-1 - Parakeet Reference Network

**Traffic Statistics provided by the Parakeet Specification (per day call initiations)**

| Src\Dest | S1 | S2 | S3 | S4 | S5 | S6 | S7 | L1 | L2 | L3 | L4 | L5 | L6 | L7 | L8 | L9 | L10 | L11 | L12 | L13 | Tot |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| S1 | 165 | 20 | | | | 40 | | 40 | 10 | | | | 10 | 5 | 30 | 20 | 30 | | | | 370 |
| S2 | 20 | 165 | 20 | | | 40 | | 20 | 40 | 20 | | | 10 | 5 | 30 | 20 | 30 | | | | 420 |
| S3 | | 20 | 165 | 20 | | 40 | | | 20 | 40 | 20 | | 10 | 5 | 30 | 20 | 30 | | | | 420 |
| S4 | | | 20 | 165 | 20 | 40 | | | | 20 | 40 | 20 | 10 | 5 | 30 | 20 | 30 | | | | 420 |
| S5 | | | | 20 | 165 | 40 | | | | | 20 | 40 | 10 | 5 | 30 | 20 | 30 | | | | 380 |
| S6 | 40 | 40 | 40 | 40 | 40 | 185 | 35 | | | | | | 30 | 35 | | | | | 35 | 15 | 535 |
| S7 | | | | | | 35 | 165 | | | | | | 10 | 5 | | | | 35 | | | 250 |
| L1 | 30 | 20 | | | | 10 | | 95 | 35 | | | | 10 | 10 | 10 | 20 | 20 | | 15 | 13 | 288 |
| L2 | 10 | 30 | | | | 10 | | 35 | 95 | 35 | | | 10 | 5 | 10 | 20 | 20 | | 15 | 13 | 308 |
| L3 | | 10 | 30 | | | 10 | | | 35 | 95 | 35 | | 10 | 10 | 10 | 20 | 20 | | 15 | 13 | 313 |
| L4 | | | 10 | 30 | | 10 | | | | 35 | 95 | 35 | 10 | 10 | 10 | 20 | 20 | | 15 | 13 | 313 |
| L5 | | | | 10 | 30 | 10 | | | | | 35 | 95 | 10 | 10 | 10 | 20 | 20 | | 15 | 13 | 278 |
| L6 | 10 | 10 | 10 | 10 | 10 | 30 | | 10 | 10 | 10 | 10 | 10 | 185 | 35 | 15 | 15 | 15 | | 35 | 25 | 455 |
| L7 | 10 | 10 | 10 | 10 | 10 | 30 | | | | | | | 30 | 160 | 10 | | | | 30 | 20 | 370 |
| L8 | 25 | 25 | 25 | 25 | 25 | 10 | | 10 | 10 | 10 | 10 | 10 | 15 | 10 | 150 | 40 | 40 | | 10 | 15 | 520 |
| L9 | 25 | 25 | 25 | 25 | 25 | 25 | | 10 | 10 | 10 | 10 | 10 | 15 | | 40 | 150 | 40 | | 10 | 15 | 535 |
| L10 | 25 | 25 | 25 | 25 | 25 | 25 | | 10 | 10 | 10 | 10 | 10 | 15 | | 40 | 40 | 150 | | | 10 | 535 |
| L11 | | | | | | | 30 | | | | | | | | | | | 165 | | | 195 |
| L12 | 15 | 15 | 15 | 15 | 15 | 15 | | 5 | 5 | 5 | 5 | 5 | | 10 | 10 | 10 | 10 | | 160 | 15 | 320 |
| L13 | 15 | 15 | 15 | 15 | 15 | 15 | | 5 | 5 | 5 | 5 | 5 | | 10 | 10 | 10 | 10 | | 15 | 160 | 330 |
| Totals | 390 | 430 | 410 | 410 | 380 | 620 | 230 | 290 | 335 | 345 | 345 | 290 | 420 | 335 | 465 | 465 | 505 | 200 | 360 | 330 | |

## Total Traffic - both directions - BUSY HOUR

| | S1 | S2 | S3 | S4 | S5 | S6 | S7 | L1 | L2 | L3 | L4 | L5 | L6 | L7 | L8 | L9 | L10 | L11 | L12 | L13 |
|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|------|
| S1 | | | | | | | | | | | | | | | | | | | | |
| S2 | 10 | | | | | | | | | | | | | | | | | | | |
| S3 | 0 | 10 | | | | | | | | | | | | | | | | | | |
| S4 | 0 | 0 | 10 | | | | | | | | | | | | | | | | | |
| S5 | 0 | 0 | 0 | 10 | | | | | | | | | | | | | | | | |
| S6 | 20 | 20 | 20 | 20 | 20 | | | | | | | | | | | | | | | |
| S7 | 0 | 0 | 0 | 0 | 0 | 17.5 | | | | | | | | | | | | | | |
| L1 | 17.5 | 10 | 10 | 0 | 0 | 0 | 0 | | | | | | | | | | | | | |
| L2 | 5 | 17.5 | 17.5 | 0 | 0 | 5 | 0 | 17.5 | | | | | | | | | | | | |
| L3 | 0 | 7.5 | 7.5 | 5 | 0 | 5 | 0 | 0 | 17.5 | | | | | | | | | | | |
| L4 | 0 | 0 | 7.5 | 17.5 | 7.5 | 5 | 0 | 0 | 0 | 17.5 | | | | | | | | | | |
| L5 | 0 | 0 | 0 | 7.5 | 17.5 | 17.5 | 0 | 0 | 0 | 0 | 17.5 | | | | | | | | | |
| L6 | 5 | 5 | 5 | 5 | 5 | 5 | 2.5 | 5 | 5 | 5 | 5 | 5 | | | | | | | | |
| L7 | 3.75 | 3.75 | 3.75 | 3.75 | 3.75 | 16.25 | 1.25 | 5 | 3.75 | 5 | 5 | 5 | 16.25 | | | | | | | |
| L8 | 13.75 | 13.75 | 13.75 | 13.75 | 13.75 | 2.5 | 0 | 7.5 | 7.5 | 7.5 | 7.5 | 7.5 | 7.5 | 2.5 | | | | | | |
| L9 | 11.25 | 11.25 | 11.25 | 11.25 | 11.25 | 6.25 | 0 | 10 | 10 | 10 | 10 | 10 | 10 | 0 | 20 | | | | | |
| L10 | 13.75 | 13.75 | 13.75 | 13.75 | 13.75 | 6.25 | 0 | 10 | 10 | 10 | 10 | 10 | 10 | 0 | 20 | 20 | | | | |
| L11 | 0 | 0 | 0 | 0 | 0 | 0 | 16.25 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | | | |
| L12 | 3.75 | 3.75 | 3.75 | 3.75 | 3.75 | 3.75 | 0 | 5 | 5 | 5 | 5 | 5 | 8.75 | 10 | 6.25 | 6.25 | 3.75 | 0 | | |
| L13 | 3.75 | 3.75 | 3.75 | 3.75 | 3.75 | 3.75 | 0 | 4.5 | 4.5 | 4.5 | 4.5 | 4.5 | 6.25 | 7.5 | 6.25 | 6.25 | 6.25 | 0 | 7.5 | |

**ATM Trunk Loading**
**calls per busy hour**

| | | |
|----|--------|---------|
| A1 | L6/S6 | 69.375 |
| A2 | S6/L8 | 233.125 |
| A3 | L6/L10 | 130.625 |
| A4 | L8/L10 | 154 |
| A5 | L9/L10 | 157.75 |
| A6 | L10/L11 | 130.125 |
| A7 | L10/L13 | 85 |
| A8 | L8/L9 | 224.375 |
| A9 | L9/L11 | 130.625 |

Applying statistical analysis

**Channel**
**Utilization**

30 number of channels (K)
233.125 mean arrival rate per hour

3.885417 mean arrival rate per minute (lambda)

3 mean call holding time in minutes

0.333333 mean service rate per minute (mu)

11.65625 utilization factor for each channel (ro)

0.388542 utilization factor for system (ro_barred)

8.66E-06 probability of no calls [P(0)]

| channels in use (N) | P(N) | Cum Dist | channels in use (N) | P(N) | Cum Dist | channels in use (N) | P(N) | Cum Dist |
|---|---|---|---|---|---|---|---|---|
| 0 | 0.0000 | 0.0000 | 16 | 0.0481 | 0.9163 | 31 | 0.0000 | 1.0000 |
| 1 | 0.0001 | 0.0001 | 17 | 0.0330 | 0.9493 | 32 | 0.0000 | 1.0000 |
| 2 | 0.0006 | 0.0007 | 18 | 0.0214 | 0.9706 | 33 | 0.0000 | 1.0000 |
| 3 | 0.0023 | 0.0030 | 19 | 0.0131 | 0.9837 | 34 | 0.0000 | 1.0000 |
| 4 | 0.0067 | 0.0096 | 20 | 0.0076 | 0.9914 | 35 | 0.0000 | 1.0000 |
| 5 | 0.0155 | 0.0252 | 21 | 0.0042 | 0.9956 | 36 | 0.0000 | 1.0000 |
| 6 | 0.0302 | 0.0554 | 22 | 0.0022 | 0.9979 | 37 | 0.0000 | 1.0000 |
| 7 | 0.0503 | 0.1056 | 23 | 0.0011 | 0.9990 | 38 | 0.0000 | 1.0000 |
| 8 | 0.0732 | 0.1789 | 24 | 0.0006 | 0.9996 | 39 | 0.0000 | 1.0000 |
| 9 | 0.0948 | 0.2737 | 25 | 0.0003 | 0.9998 | 40 | 0.0000 | 1.0000 |
| 10 | 0.1106 | 0.3843 | 26 | 0.0001 | 0.9999 | 41 | 0.0000 | 1.0000 |
| 11 | 0.1172 | 0.5014 | 27 | 0.0000 | 1.0000 | 42 | 0.0000 | 1.0000 |
| 12 | 0.1138 | 0.6152 | 28 | 0.0000 | 1.0000 | 43 | 0.0000 | 1.0000 |
| 13 | 0.1020 | 0.7173 | 29 | 0.0000 | 1.0000 | 44 | 0.0000 | 1.0000 |
| 14 | 0.0850 | 0.8022 | 30 | 0.0000 | 1.0000 | 45 | 0.0000 | 1.0000 |
| 15 | 0.0660 | 0.8682 | | | | | | |

| channels in use (N) | P(N) | Cum Dist |
|---|---|---|
| 46 | 0.0000 | 1.0000 |
| 47 | 0.0000 | 1.0000 |
| 48 | 0.0000 | 1.0000 |
| 49 | 0.0000 | 1.0000 |
| 50 | 0.0000 | 1.0000 |
| 51 | 0.0000 | 1.0000 |
| 52 | 0.0000 | 1.0000 |
| 53 | 0.0000 | 1.0000 |
| 54 | 0.0000 | 1.0000 |
| 55 | 0.0000 | 1.0000 |
| 56 | 0.0000 | 1.0000 |
| 57 | 0.0000 | 1.0000 |
| 58 | 0.0000 | 1.0000 |
| 59 | 0.0000 | 1.0000 |
| 60 | 0.0000 | 1.0000 |

# Appendix C: Call Activity Calculations

A spreadsheet was developed to determine the bandwidth used given the probability distribution of call connection activity. The probability density function derived from the traffic analysis was used in conjunction with the table of bit rate per number of connections carried which was calculated based on the AAL design. The weighted average bit rate can then be calculated. The spreadsheet is represented in the table given below:

Table C-1: Calculation of Effective Bitrate Given Parakeet Call Activity

| Chnls | TDM | Bit Rate for no. of Chnl (kbps) | | | P(N) | TDM | Dyn CES | AAL1 | AAL-CU |
| | | Dyn CES | AAL1 | AAL-CU | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 512.0 | 24.8 | 18.0 | 20.7 | 0.00 | 0.0 | 0.0 | 0.0 | 0.0 |
| 1 | 512.0 | 42.5 | 36.1 | 41.5 | 0.00 | 0.1 | 0.0 | 0.0 | 0.0 |
| 2 | 512.0 | 60.2 | 54.1 | 62.2 | 0.00 | 0.3 | 0.0 | 0.0 | 0.0 |
| 3 | 512.0 | 77.9 | 72.2 | 83.0 | 0.00 | 1.2 | 0.2 | 0.2 | 0.2 |
| 4 | 512.0 | 95.6 | 90.2 | 103.7 | 0.01 | 3.4 | 0.6 | 0.6 | 0.7 |
| 5 | 512.0 | 113.4 | 108.3 | 124.5 | 0.02 | 8.0 | 1.8 | 1.7 | 1.9 |
| 6 | 512.0 | 131.1 | 126.3 | 145.2 | 0.03 | 15.5 | 4.0 | 3.8 | 4.4 |
| 7 | 512.0 | 148.8 | 144.3 | 166.0 | 0.05 | 25.7 | 7.5 | 7.3 | 8.3 |
| 8 | 512.0 | 166.5 | 162.4 | 186.7 | 0.07 | 37.5 | 12.2 | 11.9 | 13.7 |
| 9 | 512.0 | 184.2 | 180.4 | 207.5 | 0.09 | 48.6 | 17.5 | 17.1 | 19.7 |
| 10 | 512.0 | 201.9 | 198.5 | 228.2 | 0.11 | 56.6 | 22.3 | 21.9 | 25.2 |
| 11 | 512.0 | 219.6 | 216.5 | 249.0 | 0.12 | 60.0 | 25.7 | 25.4 | 29.2 |
| 12 | 512.0 | 237.4 | 234.6 | 269.7 | 0.11 | 58.3 | 27.0 | 26.7 | 30.7 |
| 13 | 512.0 | 255.1 | 252.6 | 290.5 | 0.10 | 52.2 | 26.0 | 25.8 | 29.6 |

| | | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| 14 | 512.0 | 272.8 | 270.6 | 311.2 | 0.08 | 43.5 | 23.2 | 23.0 | 26.4 |
| 15 | 512.0 | 290.5 | 288.7 | 332.0 | 0.07 | 33.8 | 19.2 | 19.1 | 21.9 |
| 16 | 512.0 | 308.2 | 306.7 | 352.7 | 0.05 | 24.6 | 14.8 | 14.8 | 17.0 |
| 17 | 512.0 | 325.9 | 324.8 | 373.5 | 0.03 | 16.9 | 10.7 | 10.7 | 12.3 |
| 18 | 512.0 | 343.6 | 342.8 | 394.2 | 0.02 | 10.9 | 7.3 | 7.3 | 8.4 |
| 19 | 512.0 | 361.3 | 360.9 | 415.0 | 0.01 | 6.7 | 4.7 | 4.7 | 5.4 |
| 20 | 512.0 | 379.1 | 378.9 | 435.7 | 0.01 | 3.9 | 2.9 | 2.9 | 3.3 |
| 21 | 512.0 | 396.8 | 396.9 | 456.5 | 0.00 | 2.2 | 1.7 | 1.7 | 1.9 |
| 22 | 512.0 | 414.5 | 415.0 | 477.2 | 0.00 | 1.1 | 0.9 | 0.9 | 1.1 |
| 23 | 512.0 | 432.2 | 433.0 | 498.0 | 0.00 | 0.6 | 0.5 | 0.5 | 0.6 |
| 24 | 512.0 | 449.9 | 451.1 | 518.7 | 0.00 | 0.3 | 0.2 | 0.2 | 0.3 |
| 25 | 512.0 | 467.6 | 469.1 | 539.5 | 0.00 | 0.1 | 0.1 | 0.1 | 0.1 |
| 26 | 512.0 | 485.3 | 487.1 | 560.2 | 0.00 | 0.1 | 0.1 | 0.1 | 0.1 |
| 27 | 512.0 | 503.0 | 505.2 | 581.0 | 0.00 | 0.0 | 0.0 | 0.0 | 0.0 |
| 28 | 512.0 | 520.8 | 523.2 | 601.7 | 0.00 | 0.0 | 0.0 | 0.0 | 0.0 |
| 29 | 512.0 | 538.5 | 541.3 | 622.5 | 0.00 | 0.0 | 0.0 | 0.0 | 0.0 |
| 30 | 512.0 | 556.2 | 559.3 | 643.2 | 0.00 | 0.0 | 0.0 | 0.0 | 0.0 |

**effec-
tive
band-
width:** 512.0 231.3 228.3 262.6

**A Case for Call Activity Detection for ATM Voice Services in Project Parakeet**

W.D. Blair

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The introduction of Asynchronous Transfer Mode (ATM) into the Australian Army tactical communication system will provide significant benefits through responsive network management and the integration of multiple communication applications. ATM integration into Parakeet is complicated as the current network employs military standard protocols which are quite different to civil standards. This report seeks to quantify capacity economies which can be achieved through the development of an adaptive approach to voice services on the ATM network by transmitting only active connections. An examination is also made to determine whether data communications protocols can access the released capacity. The study finds that savings are significant and the released capacity should add substantially to data communications capability.