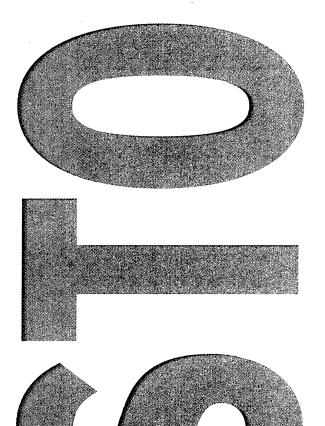**Australian Government**
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Simulation of Voice over IP for the Battlespace Communications System (Land)

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ABSTRACT

Project JP2072 aims to enhance communications in the land environment, providing more capacity for computer data traffic. One potential option is an Internet Protocol-based network carrying both voice and data. An important consideration for a converged network is whether or not acceptable voice services can be supported given the additional data traffic load. This report describes a method to estimate the quality of Voice over IP calls in the presence of other network traffic, based on the ITU E-Model. Required inputs to the E-Model are determined from a computer simulation of the network. We illustrate the application of the model to a reference Parakeet network. Two companion reports present corresponding simulations of ATM-based solutions, which, together with the simulation described in this report, will enable an evaluation of the options proposed for project JP2072.

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Simulation of Voice over IP for the Battlespace Communications System (Land)

Executive Summary

There is an increasing demand for computer data communications in the land environment, and the existing circuit-switched network infrastructure is not well suited for the carriage of such traffic. This issue is to be addressed by project JP2072, which will provide enhanced communications for land forces. This is likely to result in a move to a converged (voice and data), packet-based network infrastructure, possibly based on Voice over Internet Protocol (VoIP) or Asynchronous Transfer Mode (ATM) technology. It is expected that such solutions would provide additional bandwidth for data traffic. However, it is crucial that an acceptable quality of voice services is maintained as additional data is added to the tactical network. Hence there is a need to estimate the impact a given data traffic load will have on the quality of voice calls carried by the network, for a particular technological solution. This report describes a method for making such an assessment when voice services are implemented using VoIP on a purely IP-based network. We illustrate the method by applying it to a reference Parakeet network.

Our approach is based on the International Telecommunication Union (ITU) E-Model – a standard analytic method that allows the quality of a voice call as perceived by the caller (and rated on a scale of 0 to 100) to be estimated from objective measures relating to the network and to the terminal equipment at each end. We obtain the measures required for input to the E-Model by executing a computer simulation of the network, using the commercial network simulator OPNET. With this software a model can be created to represent any network topology, supporting any voice call and application data load. Our custom modifications to the simulator directly provide complete statistics for the E-Model transmission rating factor, for any pair of participating nodes. The simulator can also be used to determine the impact of other network characteristics on voice call quality, such as the choice of Quality of Service (QoS) policies.

Two companion papers describe a corresponding simulation for the evaluation of voice call quality in ATM-based networks. Once configured with realistic application data profiles, these simulations will enable a thorough comparison to be made between the two technologies, and will facilitate the choice of suitable network parameters in each case. This analysis will be used in the evaluation of options proposed for project JP2072.

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Contents

1. Introduction

Recently there has been considerable interest in migrating military communications infrastructure from the existing situation, where voice telephony services and computer data services are provided by separate, stove-piped networks, to a converged solution where both traffic types can exist on a single network. Using the same network for voice and data is expected to make system maintenance easier, and the use of packetised voice should make additional bandwidth available for the ever-increasing demand for data services (during the gaps between periods of speech). A number of technologies have been developed to implement packetised voice – two of the most important being based on Internet Protocol (IP) and Asynchronous Transfer Mode (ATM) – and there is a need to compare the performance of proposed solutions for voice and data integration under expected conditions of use in military operations. In general, the complexity of these networks rules out an approach based on mathematical analysis alone (for example, using queueing theory). Useful performance metrics can be measured in laboratory testbeds (such as the Application Performance Testbed for ADF Communications – APTAC – used in DSTO), but such experiments are usually restricted to small networks and produce results that are not readily generalised to larger scales. In some cases the technology to be assessed is not yet available, or the operating conditions of interest are not easily reproduced in the lab. In such situations computer simulation of the network can provide a useful tool for performance analysis. Network simulations can reproduce protocol behaviour and communications effects to any required level of detail; they can be scaled up to investigate large topologies which, realistically, are not feasible to assemble; and they can incorporate new component designs that are not yet available in real networks. In this report we describe the development of a simulation model that can be used to investigate the performance of Voice over IP (VoIP) in military networks. Two companion reports [1, 2] apply the concepts presented here to address the performance of ATM-based solutions. A fourth paper in the series compares the results obtained for each of the three proposed solutions [3].

Previous simulation studies of VoIP have focused on a number of network-level performance measures that are known to impact on call quality, namely, the end-to-end packet delay, delay variation (jitter) and packet loss between the source and destination [4, 5, 6]. However, it is difficult to make network design decisions when these three measures are considered separately. For example, will voice call quality be improved by decreasing the end-to-end packet delay, if this is at the expense of increasing the packet loss ratio? The International Telecommunication Union (ITU) has standardised the E-Model, a framework that allows a quantitative estimate to be derived for the perceived quality of a voice call, based on characteristics of the user terminals and the connection between them. The result of applying the E-Model is a rating for the call quality on a scale of 0 (worst) to 100 (best). The E-Model enables the comparison of different network configurations with respect to voice quality by combining the various network-level performance metrics into a single voice rating

factor. The E-Model has been widely used to predict the performance of existing and planned networks [7, 8, 9, 10], however, previous studies have either relied on measurements taken from existing networks to provide the necessary E-Model inputs, or else have derived approximations for the inputs based on simplifying assumptions.

For this study we have implemented the ITU E-Model in the high-fidelity network simulator OPNET™ [11]. OPNET incorporates a library of very detailed models of network devices and protocols, including VoIP and other network applications as well as the Quality of Service (QoS) policies implemented in modern routers. In our implementation the voice quality rating can be selected as an output statistic to be collected during simulation execution, in the same way as other pre-defined application performance metrics (such as web page download time, for example). We can thus use the tool to specify the network topology, configure the traffic load with a high degree of realism, and experiment with alternative QoS configurations to optimise the service provided by the network to both non-voice traffic and, with our E-Model customisations, voice traffic.

We have applied our OPNET implementation of the E-Model to a reference network appearing in early Parakeet specifications and described by Blair and Jana [12]. This reference specifies the network topology and the expected number of voice calls between each pair of nodes; both of these aspects are reproduced in our OPNET model. We have also investigated the impact of including non-voice data traffic on the network in addition to the voice traffic load. Although we do not yet have an endorsed data traffic profile, we have demonstrated the way in which such a profile will be implemented in OPNET once it is available, and the type of experiments that can be conducted with the resulting model.

The structure for the remainder of this report is as follows. Section 2 explains the ITU E-Model, and describes our implementation of the E-Model using OPNET. In Section 3 we describe the configuration of the OPNET model of the Parakeet reference network, and present the results of a number of simulation runs under different conditions. In Section 4 we discuss some of the issues arising in the application of the E-Model. Appendix A includes more detailed results for each of the scenarios simulated.

2. Predicting Voice Call Quality

2.1 The E-Model

ITU-T Recommendation G.107 specifies the algorithm for the so-called E-Model, a computational model that can be used to predict the overall quality of a voice call given a number of transmission parameters [13]. The output of the model is the voice quality 'transmission rating factor', which is interpreted as a measure of customer satisfaction on a scale of 0 (worst) to 100 (best). The standard describes how the rating

factor may be transformed into other commonly used measures of customer satisfaction, such as the Mean Opinion Score (MOS). It is stressed, however, that the model is intended to be used only for relative comparisons of transmission conditions, rather than for actual customer opinion prediction. The algorithm has as inputs a large number of parameters describing the equipment used at each end of the conversation and the conditions for the line connecting them. In effect, the E-Model provides an empirical fit to previously collected data (gained from field surveys or laboratory tests), relating these transmission parameters to user satisfaction.

De Vleeschauwer *et al.* [7] have investigated the prediction of subjective voice call quality in VoIP networks, based on the E-model. In a subsequent paper [8] they have used the E-model to determine the maximum permissible transmission delay for VoIP calls carried over satellite, for a range of codecs, in order that call quality is maintained at expected levels. They find that two geostationary satellite hops will result in unacceptable call quality for any codec, and for some codecs even one geostationary satellite hop is unacceptable. Cole and Rosenbluth [9] have refined the approach, describing a method for monitoring VoIP applications based upon a reduction of the E-model to transport-level, measurable quantities. In their application the measurable quantities (eg. delay and packet loss) are sampled from a real network in order to continuously monitor the performance of the network. We have applied their method to our simulated network, with the measurable quantities obtained as outputs of the simulation execution. In this way we can predict the performance of planned networks in carrying VoIP calls, as a tool to aid with network design.

The output of the E-model, the Transmission Rating Factor, R, is computed according to the following expression:

$$R = R_0 - I_s - I_d - I_e + A, \qquad (1)$$

where R_0 represents the basic signal-to-noise ratio; I_s is a combination of all impairments that occur approximately simultaneously with the voice signal; I_d represents the impairments due to delay; I_e represents impairments caused by low bit-rate codecs; and A allows for compensation of impairment factors when there are other advantages of access or cost to the user. Following Cole and Rosenbluth [9], we adopt default values for the first two terms (given in G.107). These terms are determined by the characteristics of the terminal equipment used to access the packet network, rather than the characteristics of the network itself. We also drop the expectation factor, A, since it is rather subjective, and in any case would take on the same value for the different options being compared. We therefore obtain[1]:

$$R = 93.3 - I_d - I_e. \qquad (2)$$

The term I_d is a function of two types of variables: those representing delays in the network, and other variables (for example, representing the level of echo suppression). The latter group of variables are assigned default values. Three variables representing

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[1] The value 93.3 appearing in our expression differs slightly from the value of 94.2 given by Cole and Rosenbluth [9] due to changes made in the year 2000 revision of the E-model.

delays are used: 1) T_a the average, absolute one-way mouth-to-ear delay; 2) T the average, one-way delay from the receive side to the point in the end-to-end path where a signal coupling occurs as a source of echo; and 3) T_r the average, round-trip delay in the four-wire loop. For the current VoIP scenario with no circuit-switched components these delays are related as

$$T_a = T = T_r / 2 ,$$ (3)

and I_d is a function, specified in G.107, of the single delay T [9]. This delay consists of three components: 1) the component due to the codec, 2) the network delay, and 3) the delay due to the de-jitter buffer[2]. The codec delay component is determined from the look-ahead time, the frame size and the number of frames per packet. The network delay component will be measured from the simulation, and includes delays due to transmission, propagation and queueing in routers. The de-jitter buffer is required to smooth out the variations in the inter-arrival times of incoming packets in order to reconstruct a synchronous bit stream for playback to the receiver. It therefore removes jitter at the expense of increased delay and packet loss (due to over-run and under-run of the buffer). Calculation of the delay and packet loss arising due to the de-jitter buffer requires knowledge of the de-jitter algorithm, which is not specified as part of the codec (it is usually proprietary).

There are no analytic expressions for the final term, the equipment impairment I_e; it must be determined experimentally for a particular combination of codec, loss concealment algorithm, packet loss distribution, etc. Some measurements for common codecs are provided in Appendix I of ITU-T recommendation G.113 [14]. Included are impairment values for the G.729 codec (Annexe A, with speech activity detection and 2 frames per packet) for a range of values for the packet loss ratio, when the packet loss is random. Impairment values provided for other codecs indicate that the impairment is greater when packet loss is correlated, so the values provided for G.729 should be considered optimistic. Another potential problem with using the impairment measurements from G.113 is that we have no way of extrapolating these measurements to the other codec variants of interest (G.729B, with or without speech activity detection and with a variable number of frames per packet).

Figure 1 shows the predictions of the E-model as a function of transmission delay for the G.729A codec, using the equipment impairment measurements in G.113.

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[2] A de-jitter buffer receives incoming packets at the receiver end and stores them for a short period before playing them out, so that voice samples are played out continuously and in order (see Section 2.2).

Figure 1: E-model predictions for G.729A codec

2.2 Implementation of the E-Model in OPNET[3]

Our OPNET implementation of the E-Model provides additional statistics (transmission rating, delay impairment and equipment impairment) that a user of the software can select for collection in the same way as other application-level statistics. These statistics are available under the 'Voice Called Party' and 'Voice Calling Party' statistics groups for each node. In the standard models supplied with OPNET only 'Voice Called Party' statistics provide separate values for each source node at the other end of the conversation; 'Voice Calling Party' statistics are aggregated across all source nodes into a single value. However, we are interested in the voice quality for each node pair (for a given node, the quality of the call will be strongly dependent on which node is at the other end). We therefore modified the OPNET process models implementing the voice application in order to obtain separate values for 'Voice Calling Party' statistics also.

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[3] We used OPNET Version 8.0.C.

In our modified models an estimate for the transmission rating factor is generated as each VoIP packet arrives at its destination, in accordance with Equation (2) (these individual samples would normally be subject to some form of averaging in an actual simulation execution). For the delay impairment we have used a simple analytic fit for the function I_d as opposed to the full expression given in G.107 [13]. We used the fit provided by Cole and Rosenbluth [9]:

$$I_d = \begin{cases} 0.024T & T < 177.3\text{ms} \\ 0.024T + 0.11(T - 177.3) & T \geq 177.3\text{ms} \end{cases} \tag{4}$$

As noted above, the delay T is made up of three components. The end-to-end network delay is determined from the simulation by subtracting the creation time of the packet (carried by the packet as a timestamp) from the simulation time at which it arrives at its destination. The codec delay is fixed at 45ms, consisting of the 5ms look-ahead time required for the G.729 codec and four frames of size 10ms each. The remaining delay component is due to the de-jitter buffer.

In order for the receiver (listener) to understand the sender (speaker), the voice packets must be played out continuously at a uniform rate. However, the voice packets do not arrive at the receiver in such a uniform fashion given that network delays are not constant. The role of a de-jitter buffer is to receive incoming packets and to store them for a short period before playing them out, in order to absorb the variations in the packet delays (and possibly to correct for packets arriving in the wrong order). The de-jitter buffer therefore removes jitter at the expense of additional delay and packet loss. To determine the impact on the perceived call quality in the model we need to implement the de-jitter buffer in OPNET. However, de-jitter buffer implementations are not standardised in the way that codecs are, and tend to be specific to a particular VoIP software implementation. Our approach has been to implement an E-policy de-jitter buffer [15] within the OPNET VoIP models, as a very simple, generic example. If necessary this implementation could be replaced with alternatives in order to assess the impact of different de-jitter buffer implementations on call quality.

The E-policy is very straightforward: the buffer is maintained at a particular size until a late packet arrives that has missed its play-out slot, at which time the buffer size is increased to accommodate the late packet. Therefore, there is no packet loss at the de-jitter buffer induced in this implementation, and the total delay T increases in a step-wise fashion such that it is always the maximum of the delays experienced by all packets up to that point in time. This total delay T is used in Equation (4) to compute the delay impairment factor.

Although no packets are discarded by the de-jitter buffer, there is still some packet loss in the network. Measured values of the equipment impairment factor for different values of the packet loss ratio, for the G.729A codec (with speech activity detection and two frames per packet), are provided in G.113 [14]; we use an analytic fit to these measurements from Cole and Rosenbluth [9]:

$$I_e \approx 11 + 40\ln(1 + 10e), \tag{5}$$

where e is the packet loss ratio. As each packet arrives at its destination we compute the packet loss ratio to be used in this expression as a moving average with a 10 second window, using sequence numbers to keep track of those packets that have not been received in the window. The window is offset from the current time by the size of the de-jitter buffer, to ensure that we don't count packets as being lost prematurely.

By combining the utility of the E-Model in predicting voice call quality with the flexibility and high-level of detail possible with network simulation, we have a very useful tool for assessing the suitability of VoIP in any conceivable network. The only potentially significant source of error is the value for the equipment impairment, having been measured for a different codec to the one used in the simulation.

3. Application to the Parakeet Reference Model

3.1 Configuration of the Parakeet Reference Model

We have applied our modified OPNET voice model to a reference Parakeet network, as specified by Blair and Jana [12]. The network consists of twenty nodes; in our simulation, each of these nodes is represented by a subnet containing a Local Area Network (LAN) model connected to a router model (Figure 2). Five of the nodes form the core of the network and the corresponding routers are connected by fibre optic links operating at 2048 kbps, as shown in Figure 3. The other links between routers are radio links: satellite links operating at 512 kbps and line-of-sight radio relay links, with between one and three hops, operating at 2048 kbps. Although these could be represented in OPNET using radio link models, we have chosen to use point-to-point link models instead since all of these links are operating in a full duplex mode and simulation of point-to-point links is much faster than that of radio links.

The voice traffic profile is also specified for the reference Parakeet network [12]. The voice traffic matrix gives the average number of calls made from any node to any other node during a 24-hour period. The expected number of calls between two nodes during the 'busy hour' is one quarter of the daily average (reproduced in Appendix A). For each node in the simulation we configure a voice application profile such that the average total number of external calls made from that node during the busy hour is consistent with the voice traffic matrix (calls within a node are not simulated). This fixes the average time between calls made from a given node; to ensure that these calls go to other nodes in the correct proportions we also specify the voice destination preferences for the node to be weighted according to the entries in the voice traffic matrix. The durations of all calls in the simulation are exponentially distributed with a mean duration of 3 minutes. All calls use the G.729 codec with four frames (each of duration 10ms) per packet and with speech activity detection disabled.
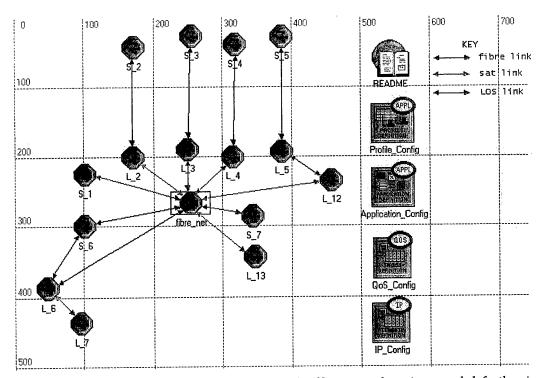
Figure 2: Top-level Parakeet network topology. The fibre_net subnet is expanded further in Figure 3 below, otherwise, each subnet consists of a LAN model and a router model.
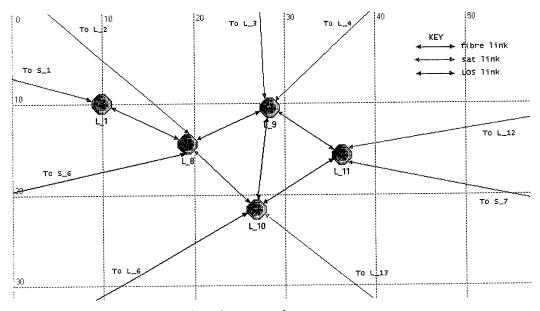

Figure 3: Fibre-connected core of Parakeet network.

As yet there is no authoritative data traffic profile for the reference Parakeet network. We have conducted simulations with web browsing of images in addition to the voice traffic in order to determine the capabilities of the simulation in regard to investigating QoS issues. We recognise that the data traffic profile we have used is probably not very realistic, but it will be straightforward to modify it in future to a more realistic profile. The same web application profile was used for all nodes, with 10 users conducting web browsing in each node. The profile attributes were chosen simply to have a noticeable impact on the voice traffic without overloading the network. Page downloads occur on average every 10 seconds while the user is browsing (which occurs for 2 minutes in every 5), and each page contains an average of 7 images, which vary in size between 2 kilobytes and 10 kilobytes. The maximum IP packet size is 1500 bytes in this model.

3.2 Results

Each set of results described below was generated by executing the simulation for 72 minutes of simulation time. Traffic profiles began after 2 minutes (providing time for the routing tables to stabilise), and result collection began after a further 10 minutes (i.e., a 10 minute 'warm up' period, followed by 60 minutes during which results were collected). As described in Section 3.1, voice traffic profiles were configured to represent the 'busy hour'.

Our initial simulation executions were directed at measuring the quality of voice calls in the reference network in the absence of any non-voice traffic (that is, the web browsing traffic profile was disabled). We find that the transmission rating is relatively constant for each pair of nodes with the same number of satellite hops (the average Transmission Rating Factors for calls between all pairs of nodes are shown in Appendix A, Table 4). With voice traffic only, delays in this network are dominated by the latency of satellite links and the Transmission Rating Factor depends only on the number of satellite hops in the path between the two nodes – taking the network delay to be 239 ms × the number of satellite hops gives R = 81.2 (0 hops), R = 63.7 (1 hop), R = 31.7 (2 hops), as shown in Table 1. In conclusion, for this codec, any satellite link in the path will result in a call quality of 'low' or 'poor', even without the delays, jitter and packet loss that would result from the introduction of non-voice traffic to the network.

The voice-only simulation was also executed without the de-jitter buffer implementation (that is, the delay induced by the de-jitter buffer was ignored). We find that the transmission rating factors were only slightly higher than those produced with the de-jitter buffer included, as shown in Table 1. This is a consequence of the fact that with only voice traffic the load on the network is relatively low so there is no packet loss and the level of jitter is very small, so the E-policy de-jitter buffer is small and introduces a small additional delay.

The main motivation in developing a simulation model of this network is to investigate issues relating to the integration of voice and data traffic (VoIP traffic on its own is

relatively simple and can be addressed analytically [12]). As discussed in Section 3.1, we chose to add a data traffic profile based on web browsing of large images. As additional data traffic impacts the quality of voice services, and itself has different measures of service quality (such as page download time), it is necessary to configure routers with appropriate policies for QoS in order to simultaneously meet the requirements of all traffic types. However, as a baseline case, we have executed the simulation with the web browsing profile enabled and with the default router model implementing a First In, First Out packet queue of infinite length, and with no differentiation between VoIP and HTTP packets. This is to be compared with the results when QoS features of routers are enabled.

The results for the baseline case without QoS, given in Table 1, show that the call quality is significantly decreased with the addition of data traffic, for those nodes separated by at least one satellite hop. This is due to the additional delays experience by VoIP packets as they queue in the routers, given the additional data traffic. The router model used in this execution has an infinite packet buffer, so these delays can grow to be very large. In fact, the average transmission rating calculated for nodes separated by two satellite hops is negative, indicating that the delays exceed the range for which the E-model can be validly applied. The results also show that the Transmission Rating Factors vary significantly more than when only voice traffic is present (Table 5 in Appendix A shows the average Transmission Rating Factors for calls between all pairs of nodes when data traffic is included).

The results for the baseline case indicate that the additional data traffic seriously degrades the quality of voice services. We wish to determine the improvement that can be gained when the routers are configured with an appropriate QoS policy. The scheme implemented in this model was Priority Queuing. Priority Queuing gives absolute higher priority to the packets with a higher Type of Service (ToS) value, which in our case is voice, than to those with a lower ToS value (eg. HTTP). This means that each router will deliver HTTP packets only if there are no voice packets waiting in its queue. We therefore expect the Priority Queuing scheme to significantly improve the quality of voice services at the expense of increasing the delays for HTTP services.

The results shown in Table 1 illustrate the improvement achieved by implementing Priority Queuing in the routers. The Transmission Rating Factors are much closer to the values obtained when no data traffic is present (although they are still lower – the Priority Queuing scheme does not pre-empt packets that are already being transmitted, so voice packets can still experience some queueing delays). The variations in the transmission ratings are also lower, as can be seen in Table 6 of Appendix A.

Table 1: Summary of the average of the Transmission Rating Factors over all node-pairs with 0, 1 and 2 satellite hops, for each of the scenarios investigated.

Scenario	0 Satellite Hops	1 Satellite Hop	2 Satellite Hops
No de-jitter buffer (voice only)	81.2	63.7	31.7
De-jitter buffer included (voice only)	81.2	63.1	30.8
Voice + HTTP traffic (no QoS)	80.3	25.1	-21.6
With Priority Queueing	80.9	58.9	24.1

The results in Table 2 show that the page response time for the web browsing traffic is not significantly increased by the inclusion of Priority Queueing. This result will, however, be dependent on the nature of the traffic profiles supported by the network. A more realistic traffic profile may favour a different QoS scheme.

Table 2: The sample mean and standard deviation of the web traffic's page response time with and without PQ.

Page Response Time	Sample mean (sec)	Standard Deviation (sec)
No QoS	5.2	1.2
With PQ	5.3	1.3

4. Conclusions

We have incorporated the ITU E-Model for quantifying voice call quality into network simulations, allowing us to assess the suitability of proposed networks for the transport of integrated voice and non-voice data traffic. The high level of detail in these simulations means we can make credible predictions for measures of service quality for networks still in the design stage.

The E-Model allows voice quality predictions to be made for a wide range of values of some parameters, end-to-end delay being one example. However, the equipment impairment factor has not been reduced to a function of codec attributes and must still be measured in laboratory trials for any combinations of interest. Particular factors that influence the equipment impairment factor include the choice of codec; whether or not speech activity detection is enabled; the number of frames per packet; the packet loss concealment algorithm; and the statistical nature of the packet loss occurring in the network. There are a very large number of combinations for these factors, and equipment impairments have been measured for only a small fraction of these. An alternative to subjective listening tests is the ITU Perceptual Evaluation of Speech Quality (PESQ) algorithm [16], which compares the degraded speech at the destination with the reference speech as input at the source to compute a Mean Opinion Score (MOS) value. Although this allows an assessment of voice quality for codec combinations that have not been tested in laboratory trials, the algorithm is not practical for use in simulated networks. In our case it is necessary to choose the

impairment measurement for the codec configuration that most closely matches the case of interest, and hope that it provides a good estimate of the true impairment.

Using the reference Parakeet network we have illustrated how such simulations can be used to investigate different network configurations carrying integrated voice and non-voice data. As an example application we determined the improvement in call quality achieved when Priority Queueing is used in network routers as opposed to a simple First In First Out queue. We found that in this example the transmission rating factor is determined largely by the number of satellite hops between source and destination, falling sharply with each satellite hop due to the additional propagation delay. Although the E-Model (with its strong dependence on delay) is appropriate for assessing commercial telecommunications networks, where users would certainly be distracted by such delays, it may not be an adequate measure of utility for military users who have different expectations and who can use doctrine to accommodate the delay. It may be possible to develop a militarised version of the E-Model, but this would probably require a substantial effort in laboratory trials. In the meantime, the E-Model provides the only framework available to predict voice call quality from simulation results alone.

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Appendix A: Voice Traffic Matrix

Table 3: Average number of (external) calls between nodes during the busy hour

Src\Dest	S1	S2	S3	S4	S5	S6	S7	L1	L2	L3	L4	L5	L6	L7	L8	L9	L10	L11	L12	L13	Total
S1		5				10		10	2.5				2.5	1.25	7.5	5	7.5				51.25
S2	5		5			10		5	10	5			2.5	1.25	7.5	5	7.5				63.75
S3		5		5		10			5	10	5		2.5	1.25	7.5	5	7.5				63.75
S4			5		5	10				5	10	5	2.5	1.25	7.5	5	7.5				63.75
S5				5		10					5	10	2.5	1.25	7.5	5	7.5				53.75
S6	10	10	10	10	10		8.75	2.5	2.5	2.5	2.5	2.5	7.5	8.75							87.5
S7						8.75												8.75	3.75		21.25
L1	7.5	5				2.5			8.75				2.5	2.5	2.5	5	5		3.75	3.25	48.25
L2	2.5	7.5				2.5		8.75		8.75			2.5	1.25	2.5	5	5		3.75	3.25	53.25
L3		2.5	7.5			2.5			8.75		8.75		2.5	2.5	2.5	5	5		3.75	3.25	54.5
L4			2.5	7.5		2.5				8.75		8.75	2.5	2.5	2.5	5	5		3.75	3.25	54.5
L5				2.5	7.5	2.5					8.75		2.5	2.5	2.5	5	5		3.75	3.25	45.75
L6	2.5	2.5	2.5	2.5	2.5	7.5		2.5	2.5	2.5	2.5	2.5		8.75	3.75	3.75	3.75		8.75	6.25	67.5
L7	2.5	2.5	2.5	2.5	2.5	7.5							8.75		3.75	3.75	3.75		7.5	5	52.5
L8	6.25	6.25	6.25	6.25	6.25	2.5		5	5	5	5	5	3.75	2.5		10	10		3.75	3.75	92.5
L9	6.25	6.25	6.25	6.25	6.25	6.25		5	5	5	5	5	3.75	2.5	10		10		3.75	3.75	96.25
L10	6.25	6.25	6.25	6.25	6.25	6.25		5	5	5	5	5	3.75	2.5	10	10			3.75	3.75	96.25
L11							7.5														7.5
L12	3.75	3.75	3.75	3.75	3.75	3.75		1.25	1.25	1.25	1.25	1.25	3.75	3.75						3.75	40
L13	3.75	3.75	3.75	3.75	3.75	3.75		1.25	1.25	1.25	1.25	1.25	6.25	3.75					3.75		42.5
Total	56.25	66.25	61.25	61.25	53.75	108.8	16.25	48.75	60	62.5	62.5	48.75	58.75	43.75	78.75	78.75	88.75	8.75	50	42.5	

Appendix B: Transmission Rating Factor Results

Table 4: Transmission Rating Factor for each node pair, averaged over busy hour (voice traffic only). The shading of cells indicates the number of satellite hops in the path between the two nodes (or that the two nodes never call each other). Cells with no values indicate that no calls were recorded for the node pair during the simulation run.

Src\Dest	S1	S2	S3	S4	S5	S6	S7	L1	L2	L3	L4	L5	L6	L7	L8	L9	L10	L11	L12	L13
S1		62.9						81.2	63.1				81.1	63	81.2	81.1	81.1		63.1	63.2
S2	62.9		30.2										62.8	30.6	63.1	63.2	63		63.1	31
S3		30.5		30.4		63				30.6				30.6	63	63.1	63		30.8	30.9
S4			30.6		30.4	63					30.7			30.6	63.1	63.1	63.1		31	31.1
S5				30.7		62.7						30.8			62.9	62.9	62.9		30.9	30.8
S6	81.1	63	62.9	62.8	62.8		81.1	81.2	63.2	63.2	63.1	62.8	81.2	63.3	62.9	62.9	62.9		63.2	63.2
S7													81.1					81.2		
L1	81.1	63.1							63.4				63.1	63.1	81.2	81.2	81.2		63.1	63.2
L2	63.2	81.2						63.3					30.9	30.7	63.1	63.2	63.3		30.8	31
L3		30.9	81.2						30.8		31		30.8	30.9		63.3	63.4		31	30.9
L4			30.9	81.2						30.9		30.8			63.3	63.3	63.4		30.9	31.1
L5				30.8	81.2						30.8		63		63.1	63.2	63.3		81.2	30.8
L6	81.1	62.9	62.8	62.9	62.9	81.2		81.1	63.1	63	63.1	63		30.8	63.1	63.2	63.3		63.2	63.2
L7	62.9	30.7	30.6	30.8	30.7	63.3		63.1	30.9	30.8	30.8	30.7	63.4		81.2	81.2	81.2		30.7	31
L8	81.2	63.1	63.2	63.1	63	81.2		81.2	63.3	63.2	63.2	63	81.2	63.2		81.2	81.2		63.2	63.3
L9	81.1	63.1	63.2	63.2	63.1	81.2		81.2	63.3	63.4	63.3	63.2	81.2		81.2		81.2		63.4	63.3
L10	81.1	63.1	63.1	63.1	63	81.1		81.2	63.3	63.2	63.3	63.2	81.2		81.2	81.2			63.4	63.4
L11							81.2													
L12	62.9	30.7	30.6	30.8	30.8	63			30.9	31	31.1	81.2	81.2	30.7	63.2	63.3				31.1
L13	63.1	30.8	30.9	30.8	30.8	63.1			31.1	30.9		30.8		30.9	63.4	63.4	63.3		31	

Legend: No calls | 0 satellite hops | 1 satellite hop | 2 satellite hops

Table 5: Transmission Rating Factor for each node pair, averaged over busy hour (voice and web browsing, with no QoS).

Src\Dest	S1	S2	S3	S4	S5	S6	S7	L1	L2	L3	L4	L5	L6	L7	L8	L9	L10	L11	L12	L13
S1		41.9				79.8		80.3	51.8						80.1	80.3	79.9		35.1	49.4
S2	26.1		-45			34.2		19.4	80.6				43.1	7.02	37.6	-5.2	26.8		-38	-14
S3		-3.5		-12		-0.7			-7.8	80.5			-61	-9.4	11.4	11.7	18.3		-69	-36
S4			-40		-37	14.6				-3	80.5		49.8	2.88	21.7	31.5	17.6		-29	-13
S5				-42		-0.2					-22	80.7	6.75		-15	3.5	12.3		80.4	-60
S6	80	37.1	27.9	24.3	13.6		79.9	80.1	34.1	1.88	35.7	-6.4	80.6	49.5						
S7						79.6														
L1	80.6					80.2			26.6				80	52	80.3	80	80			
L2	5.32	51.8						31.4					10.4	4.42	35.1	35.1	32			
L3		80.8	80.6						-16		-16		32	1.07	6.63	41.2	10.3			
L4			-10	80.8						-32			18.1	-20	18.3	7.96	21.6			
L5				-29	80.6	23.6					-49	-25	9.5	-49	14.2	-8.7	-29	80.6		
L6	79.7	40.6	12.2	33.5		80.6		80.1	25.6	25.1	32.3	-20		51.1	80.3	80			10.8	51.2
L7	44.5	4.23	7.63	-7.5	1.73	45		44.7	2.58	-31	2.31	-56	49.4		80.3				-31	14.6
L8	80.2	42.7	42.9	37.9	19.7	80.3		80.5	39.3	24.1	37.8	22.9	80.4	50		80.1	80.7		-33	47.6
L9	80.2	46	33.8	37.9	18	79.9		80.3	30	22	38.6	5.13	80.6		80.2		80.5		39.6	52.1
L10	80	30.6	22.2	33.7	-3.3	80.4		80.2	30	22.7	30.2	20	80.5		80.4	80.4			28.6	49.6
L11							80.7													
L12	29.4	-37	-49	-17	80.5	-4.6				-43	-50	80.7		-47	47	25.9				-30
L13	49	10.6	-16	-21	-5.3	48.4			-8.7	11.3				15	49.6		53		-39	

Legend: No calls | 0 satellite hops | 1 satellite hop | 2 satellite hops

Table 6: Transmission Rating Factor for each node pair, averaged over busy hour (voice and web browsing, with Priority Queueing).

Src\Dest	S1	S2	S3	S4	S5	S6	S7	L1	L2	L3	L4	L5	L6	L7	L8	L9	L10	L11	L12	L13
S1		58.3				80.7		81	58.8				80.7	57.5	80.9	81.8	80.7			
S2	58		23.3			58.8		59.1	81	24.2			58.2	23.3	59.4	58.7	58.7			
S3		23.3		25		58.1			23.8	81	24.4		58		58.7	59.1	58.8			
S4		24.2	24		23	58.1				24.5	81	23.5		23.7	58.9	59.5	58.8			
S5				22.1		56.9					23.6	81	57.4	22	57.2	58	58.1		80.9	
S6		58.7	58	58.2	56.8		80.6					57.4	81	59.6						
S7												81		57.9						
L1	81	58.5				80.6			59.5				80.7	58.6	81	80.9	80.9		58.2	59
L2	58.7	81				80.9		59.7		24.4			80.7	23.9	60.1	59.6	59.5		24.1	24.8
L3		23.6	81			59.6			24.4		25.3		59.2	24.2	60.3	60.1	59.6		24.2	24.4
L4			24.4	81	23.6	59.5				25		24	58.9	23.8	59.5	60.2	59.8		24.8	25
L5				23.4	81	58.7	81				24.1		58.8	23.2	58.1	58.7	59.3	81	81	23.9
L6	80.6	60	58.8	58.9	57.1	57.3		80.8	58.8	59.2	59	58.4		60.1	80.9	80.9	81		58.7	60.2
L7	57.3	23.6		24.5	22.6	81.1		58.1	24.8	59.5	23.8	23.3	60.1		59.4				23.9	24.8
L8	80.9	59.6	58.8	58.9	57.2	59.6		81.1	60.2	60.3	59.5	58.1	60.2	59.4		81	81		59.4	59.6
L9	80.7	58.7	59.5	59.6	58.7	81		80.9	59.6	59.5	60.2	58.9	80.9		81				59.5	60.2
L10	80.8	59	58.9	59	57.9	80.9		80.9	59.9		59.4	58.7	80.9		81				59.7	60.3
L11																				
L12	57.3	23.6	23.6	24	80.9	58.3		58.8	23.9	24.6	25.5	81		23.6	58.9	58.9	59.4			24.8
L13	58.4	24.1	24.1	24.1	24.7	59			24.5		24.5	23.9	81		59.5	59.5	59.6		24.8	

No calls   0 satellite hops   1 satellite hop   2 satellite hops

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19. ABSTRACT
Project JP2072 aims to enhance communications in the land environment, providing more capacity for computer data traffic. One potential option is an Internet Protocol-based network carrying both voice and data. An important consideration for a converged network is whether or not acceptable voice services can be supported given the additional data traffic load. This report describes a method to estimate the quality of Voice over IP calls in the presence of other network traffic, based on the ITU E-Model. Required inputs to the E-Model are determined from a computer simulation of the network. We illustrate the application of the model to a reference Parakeet network. Two companion reports present corresponding simulations of ATM-based solutions, which, together with the simulation described in this report, will enable an evaluation of the options proposed for project JP2072.