# Analyzing Mission Critical Voice over IP Networks

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# Organization

- What is Mission Critical Voice?
- Why Study Mission Critical Voice over IP?
- Approach to Analyze Mission Critical Voice over IP
- Voice Quality Analysis (E-Model Optimization)
- Latency Analysis
- Conclusions
- Future Work

# What is Mission Critical Voice?

- Voice Communications that has the potential to cause injury or loss of life if not received properly.
- Examples of Mission Critical Voice:
  - Air Traffic Control
  - Military Communications
  - E-911 (Emergency) Services
- What components make up Mission Critical Voice
  - Acceptable Voice Quality
  - Acceptable Latency
  - High Availability/Survivability
- This research focuses on Voice Quality and Latency.

# Why Study Mission Critical Voice over IP?

- Many public and private entities are struggling with separate voice and data networks.
- Integrated networks are easier and less costly to maintain.
- Mission Critical Voice has different requirements than ordinary voice communications.
- Military and other Government Agencies (like the FAA) are researching the integration of their mission critical networks.
- Examples uses include Military Communications, Air Traffic Control Communications, Natural and Man-made Disaster Recovery.

# Approach to Analyzing Mission Critical VoIP

1. Voice Quality Analysis
   – E-Model Optimization
2. Latency Analysis
   – Simulation Model
3. Survivability and Availability Analysis
   – A topic for future research

These analyses interact with each other. Therefore, it may be necessary to run each analysis multiple times.

# Voice Quality Analysis
# The E-Model Optimization

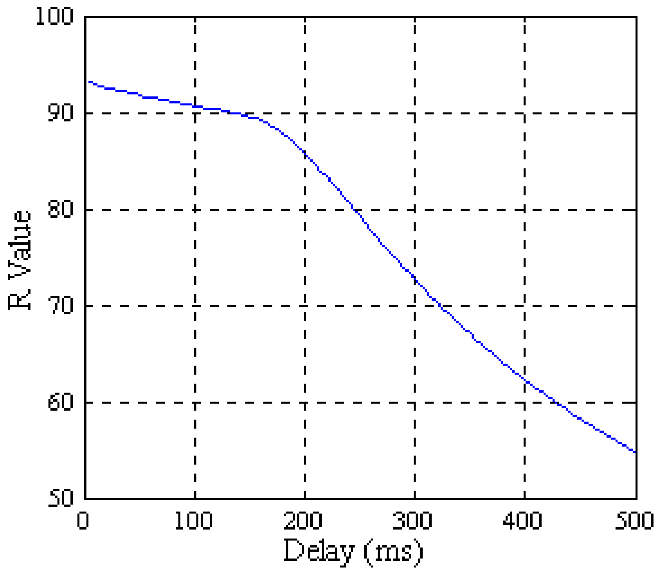
- The European Telecommunications Standards Institute (ETSI) developed the E-Model to address the needs of network planners.
- The E-model is based on the premise that "Psychological factors on the psychological scale are additive".
- The ITU and the TIA have recommended the E-Model for use.
- The E-Model defines the *R* value as the measure of voice quality.
- Comparison of Mean Opinion Score (MOS) and *R* value.

| User Satisfaction | E-model - R | MOS |
|---|---|---|
| Very Satisfied | 90 | 4.3 |
| Satisfied | 80 | 4.0 |
| Some Users Dissatisfied | 70 | 3.6 |
| Many Users Dissatisfied | 60 | 3.1 |
| Nearly All Users Dissatisfied | 50 | 2.6 |
| Not Recommended | 0 | 1.0 |

# E-Model *R* Value

$$R = Ro - Is - Id - Ie + A$$

R value vs. Delay




*Ro:* Basic signal-to-noise ratio.

*Is:* Impairments associated with voice signals, like incorrect loudness levels, quantization noise, and incorrect sidetone levels.

*Id:* Impairments associated with delay, including end-to-end and echo.

*Ie:* Equipment related impairments associated with specific equipment.

*A:* Represents an advantage factor based on “advantage of access”.

# Optimization Problem

Objective Function (all cases):

**Maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (*R*).**

The cases considered are:

1. Optimization: Find voice coder given link bandwidth, packet loss level, and link utilization.
2. Optimization: Find voice coder and packet loss level given link bandwidth and link utilization.
3. Optimization: Find voice coder and link utilization level given link bandwidth and packet loss level.

# Optimization Algorithm

1. The *Set* of system configurations is defined. For Example, In Case 2, the *Set* is the combination of coders and packet loss percentages. This allows the algorithm to search the “universe” of possibilities.
2. The parameters are calculated, including all E-Model parameters with fixed inputs and variable inputs based on the *Set*.
3. The objective is to maximize the number of calls on a link.
4. The first constraint is that the minimum R value (voice quality) is 70.
5. The second constraint is that the sum of the variable *Portion* is 1.0.

- Number of calls will be maximized with one of the *Set* combinations, Therefore this problem is an "assignment" type optimization.

# Optimization Algorithm

- AMPL Optimization Software is used to implement algorithm.
- Non-Fixed Parameters are: *T*, *Ta*, and *Tr* (Delay variables), *Ie* (Equipment Impairment), *PL* - Packet Loss %, ρ - Link Utilization, Coder Type
- T (mean one way delay of the echo path), Ta (absolute delay in echo free conditions), Tr (round trip delay in a 4-wire loop). We assume that the echo cancellers are very good. Therefore: T = Ta = (1/2)Tr.
- The variable *Portion* is used to assign the calls to a particular combination in the *Set*. Strict assignment would require *Portion* to be a binary integer (1 or 0). To avoid this non-linearity, we relaxed the integer requirement and allowed the program to make fractional assignments. Assignment theorem ensures that the solution produced will always exhibit an assignment of 1 or 0 for every *Portion* variable.

# Optimization Algorithm

- The *Code_Feas* variable is a binary variable that penalizes elements of the set that do not meet the constraints and limits the working set to $R \geq 70$.
- The algorithm can switch *Code_Feas* for that coder from a 1 to a 0 which eliminates the impairment portion of the equation and satisfies the constraint. Setting *Code_Feas* to zero for that coder eliminates it from participating in the objective which removes that coder from its working set. This is a hard (binary) penalty function that is nonlinear.
- During the first attempt, the optimization would not attempt to set the *Code_Feas* to "1" on all variables. Being non-linear, the algorithm found one coder that met the constraints and did not look for others that could produce a better objective function. This problem was solved by setting all *Code_Feas* variables to "1" during program initialization. For the algorithm to meet the $R \geq 70$ constraint, it MUST look at all *Code_Feas* variables and reverse them if necessary.

# Assumptions for Optimization

## Delay is based on M/M/1 Assumptions

$$S(X) = 1 - e^{-\mu(1-\rho)Td}$$

$$Td = \frac{\ln(PL\%)}{-\mu(1-\rho)}$$

*Ta = Hop Count*Td + Code Delay + Propagation Delay + Misc. Delay*

- *S(X)* represents M/M/1 delay distribution
- *Td* is the delay calculated by rearranging the delay distribution and assuming that the tail of the distribution is lost to packet loss (*%PL*)
- *Ta* is total one way delay

# Assumptions for Optimization

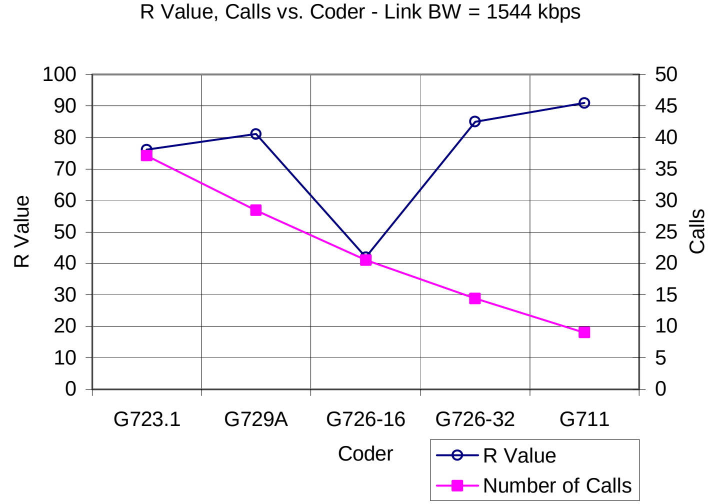
*Ie* values for different coding schemes

| Packet Loss % | G.711 with PLC Random Packet Loss (*Ie*) | G.729A + VAD (*Ie*) | G.723.1 + VAD (6.3 kbits/s) (*Ie*) |
|---|---|---|---|
| 0 | 0 | 11 | 15 |
| 0.5 | | 11 | 15 |
| 1 | 5 | 15 | 19 |
| 1.5 | | 17 | 22 |
| 2 | 7 | 19 | 24 |
| 3 | 10 | 23 | 27 |
| 4 | | 26 | 32 |
| 5 | 15 | | |
| 7 | 20 | | |
| 8 | | 36 | 41 |
| 10 | 25 | | |
| 15 | 35 | | |
| 16 | | 49 | 55 |
| 20 | 45 | | |

- *Ie* values reported in ITU G.113
- Used polynomial fit to estimate *Ie* values for packet loss % levels that were not reported in ITU G.113.

# Optimizing for Coder

- Link: 256 kbps
  - Objective: 4.3 calls
  - Coder: G.729A
- Link: 1.544 Mbps
  - Objective: 37.1 calls
  - Coder: G.723.1




- All cases were run for two different link speeds
- G.723.1 is a more efficient but offers a lower quality of voice. The significant decline in the *R* value of G726-16 as compared to the other coding schemes demonstrates the weakness of ADPCM compared to other low bit rate coding schemes .

# Optimizing for Coder, Packet Loss

Link: 256 kbps

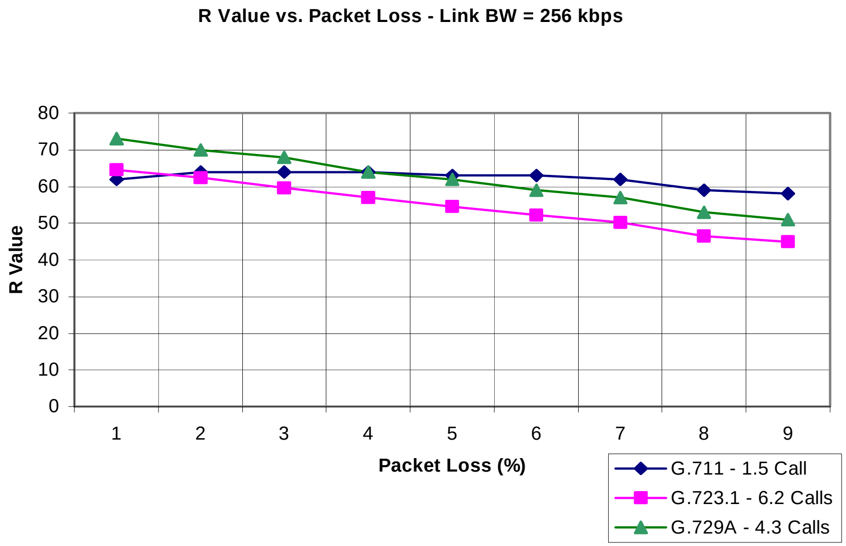
Objective: 4.3 calls

Coder: G.729A with 2% packet loss

Link: 1.544 Mbps

Objective: 37.1 calls

Coder: G.723.1 with 1% packet loss

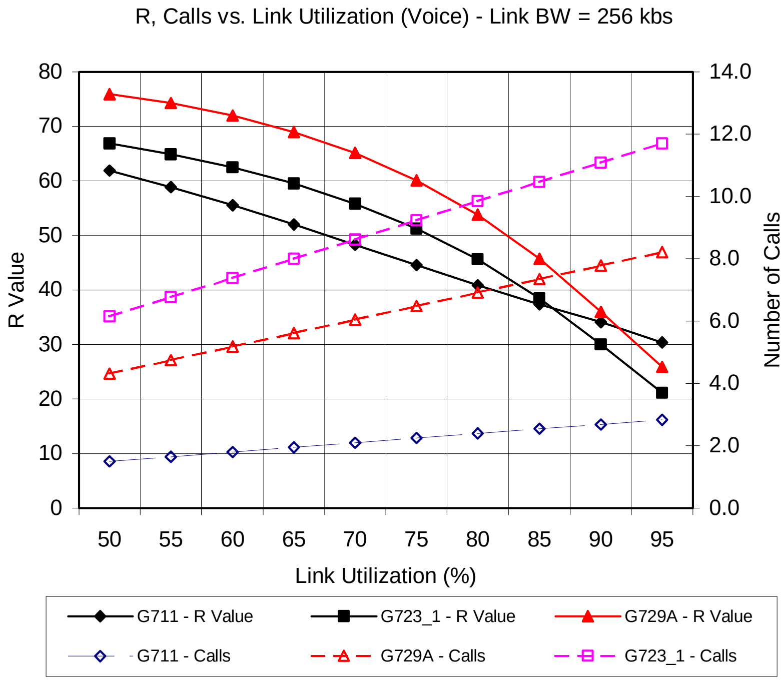


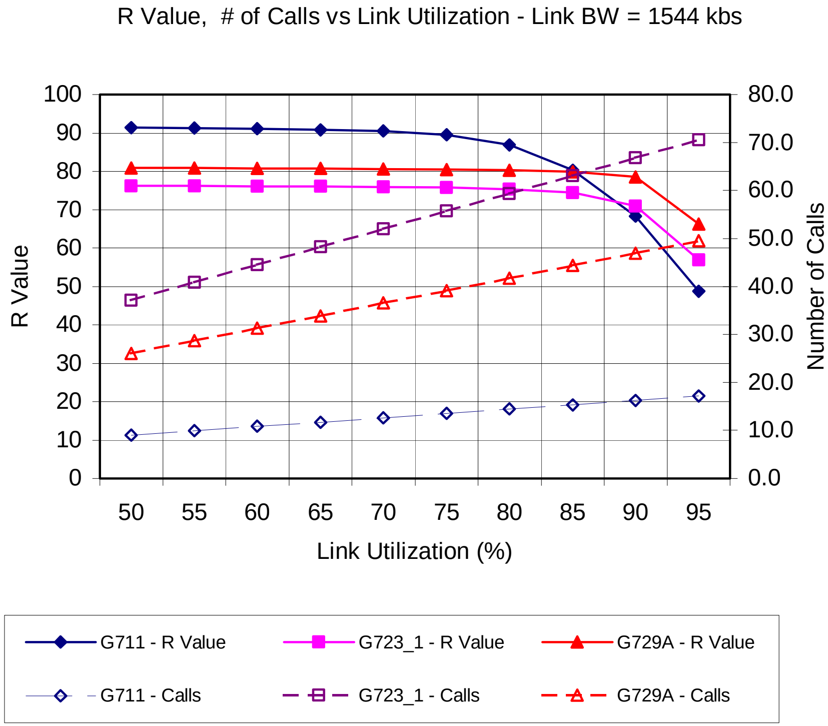

- Algorithm used a small objective benefit to favor links with higher packet losses (all other thing being equal).
- Degradation from packet loss to the audio (via the *Ie* factor) far outweighs any gains made from reductions in delay (at least using M/M/1 assumptions). This is due to the relatively steep *Ie* curves.

# Optimizing for Coder, Utilization

- Link:  256 kbps
  - Objective:  5.6 calls
  - Coder:  G.729A with 60% link utilization
- Link: 1.544 Mbps
  - Objective: 66.8 calls
  - Coder:  G.723.1 with 90%
- Notice that the *R* curves when the link speed was 256 kbps were much more sensitive than the 1.544 Mbps case.  This is due to the impact that the lower bandwidth has on delay.
- Notice that in the 1.544 Mbps case, G.723.1 carried more calls at 90% load than G.729A did at 95% load.

# Optimizing for Coder, Utilization



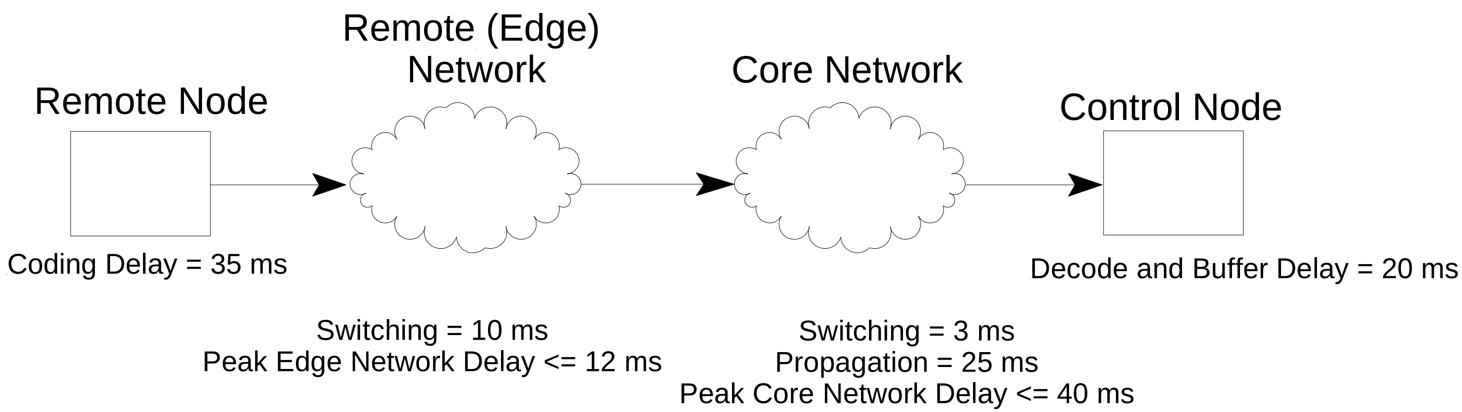
# E-Model Optimization Results

| Case # | Variables | Link Bitrate (b/s) | Optimum Solution |
|---|---|---|---|
| 1 | Coder | 256000 | G.729A |
| 1 | Coder | 1544000 | G.723.1 |
| 2 | Coder, Packet Loss % | 256000 | G.729A with 2% PL |
| 2 | Coder, Packet Loss % | 1544000 | G.723.1 with 1% PL |
| 3 | Coder, Link Utilization | 256000 | G.729A with 60% Load |
| 3 | Coder, Link Utilization | 1544000 | G.723.1 with 90% Load |

- G.729A is a better coder on lower bitrate links than G.723.1. This is due to the fact that G.729A has higher quality of voice, but is less efficient with respect to bandwidth than G.723.1.
- Packet loss typically hurt voice quality more than the delay saved (using M/M/1 assumptions).
- In the optimization for coder/load scenario, we saw that when the link BW was 1.544 Mbps, G.723.1 carried more calls at a load of 90% than G.729A did at a load of 95%. It would be interesting to run the optimization with priority queue assumptions instead of M/M/1.

# Latency Analysis - Simulation Study

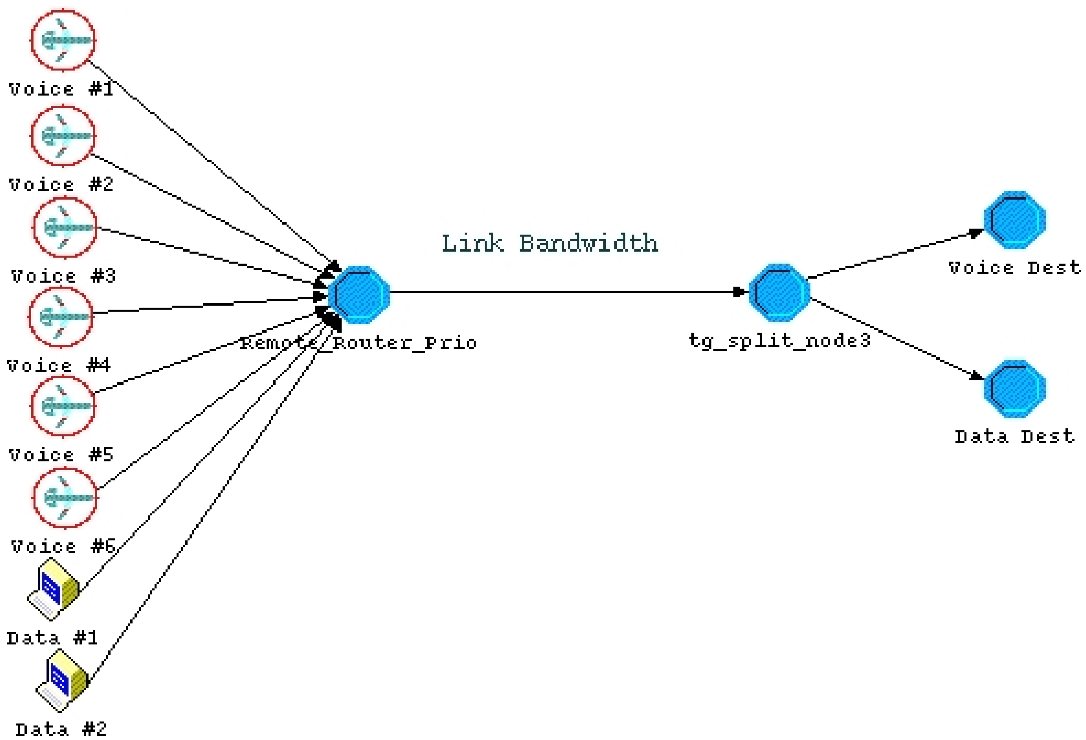
- The Remote Network model simulates the edge of the network where the voice traffic encounters a router connected to a link with limited bandwidth. The Core Network model simulates the core of the network. In the ATC environment, this would be the network between the Remote Network and the terminal or enroute Air Traffic Control facility.



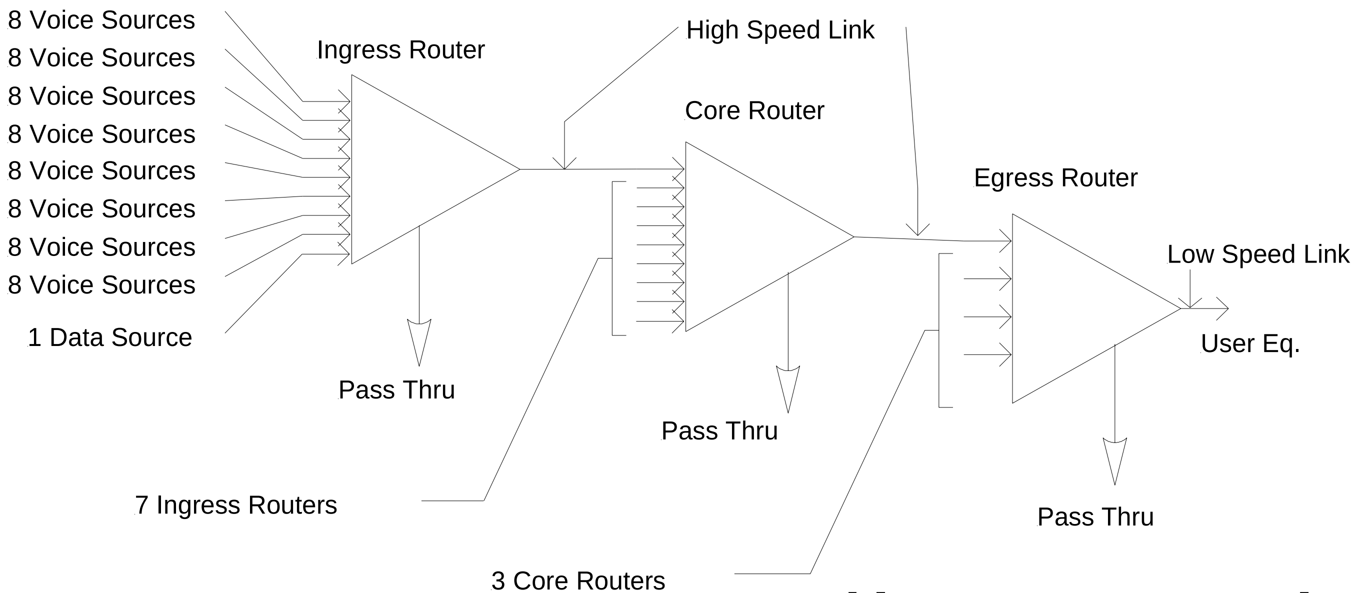

- Based on research that connects delays over 150 ms with “step on” occurrences, 145 ms was chosen as the maximum allowable delay.

# Remote Network Model

- 6 Voice Sources (on-off type)
- 2 Data Sources
- Variable Data Source Load and MTU size distribution
- Priority and DRR Queues
- Variable Link Bandwidth
- Opnet 6.0/7.0 used to build simulation models



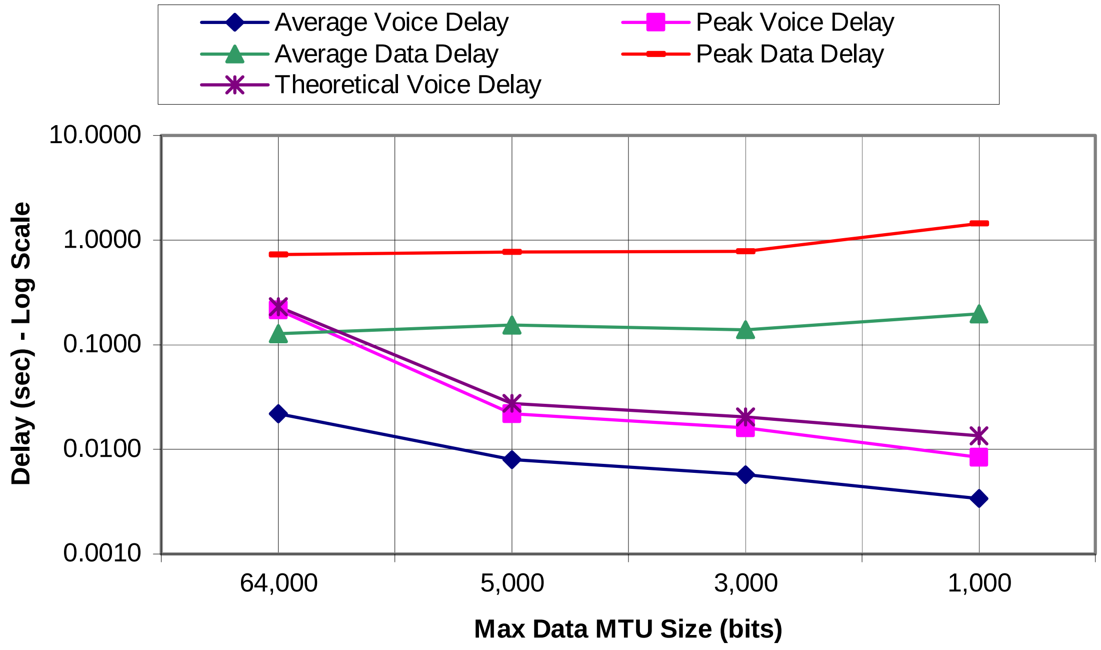
# Core Network Model




- 2048 Voice Sources, 32 Data Sources, Variable Data Source Loads and MTU size distributions
- Ingress, Core, Egress Routers – Low/High and High/Low bandwidth Transitions, Priority and DRR Queues, Variable Link Bandwidths

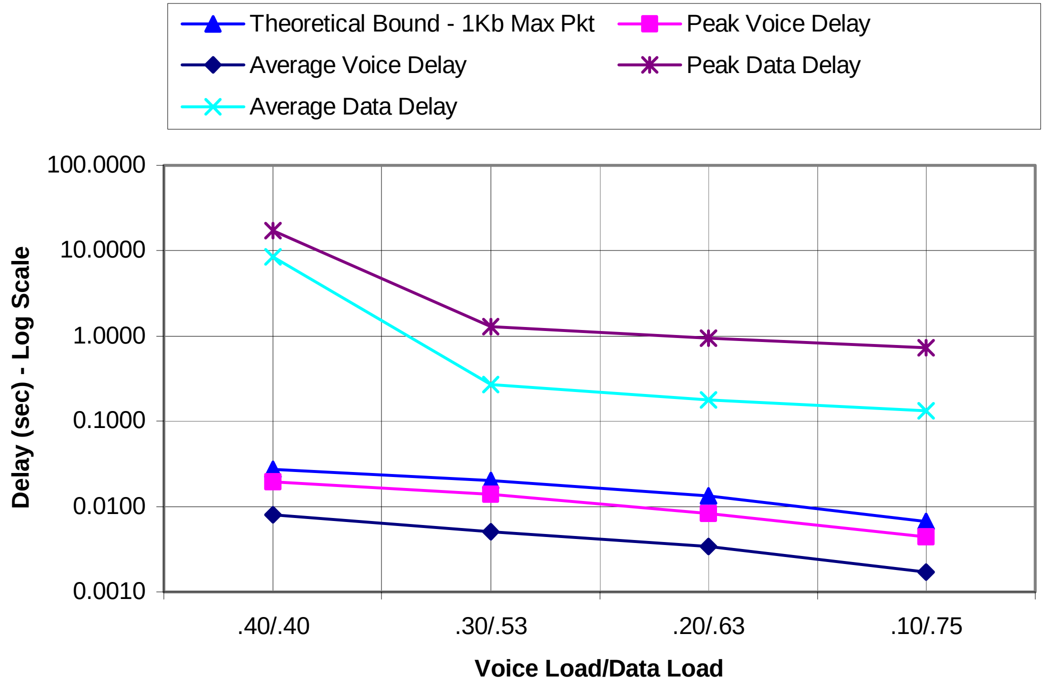
# Remote Network – MTU Size

- Data load of .64, voice load of .20  - Constant
- Link speed of 288 kbps – Constant
- Data MTU size variable



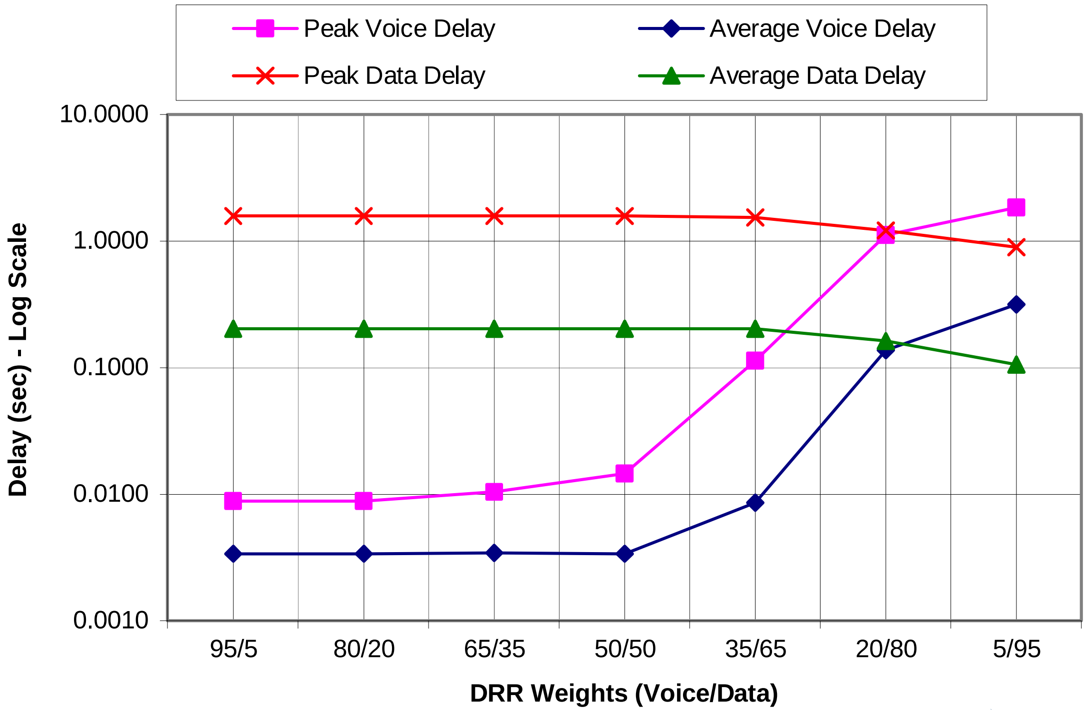
# Remote Network – Load

| Link Rate (bps) | Data Interarrival Time | Total Load | Voice Load | Data Load |
|---|---|---|---|---|
| 144,000 | 0.3203 | 0.80 | 0.40 | 0.40 |
| 192,000 | 0.1836 | 0.83 | 0.30 | 0.53 |
| 288,000 | 0.1016 | 0.83 | 0.20 | 0.63 |
| 576,000 | 0.043 | 0.85 | 0.10 | 0.75 |

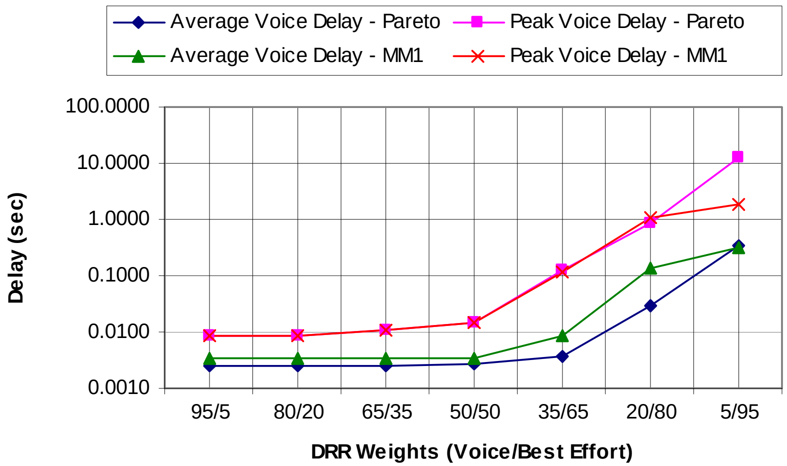


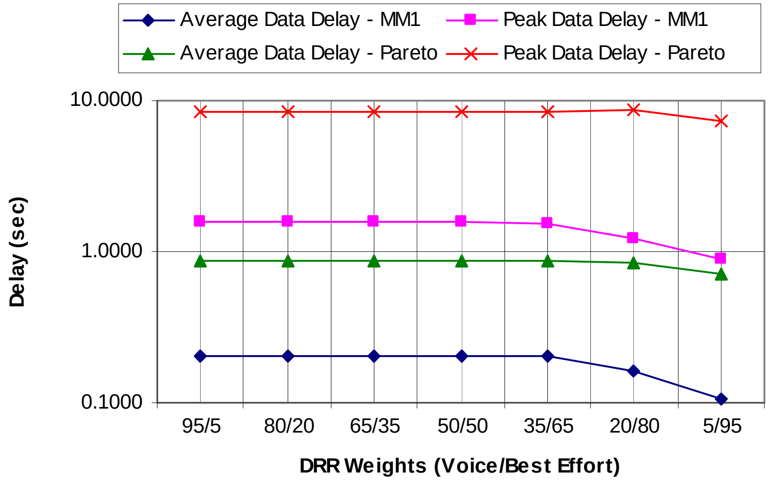
# Remote Network – DRR Queue

- Data load of .64, voice load of .20 - Constant
- Link speed of 288 kbps – Constant
- Maximum Data MTU size 1000 bits



# Remote Network – Pareto w/ DRR

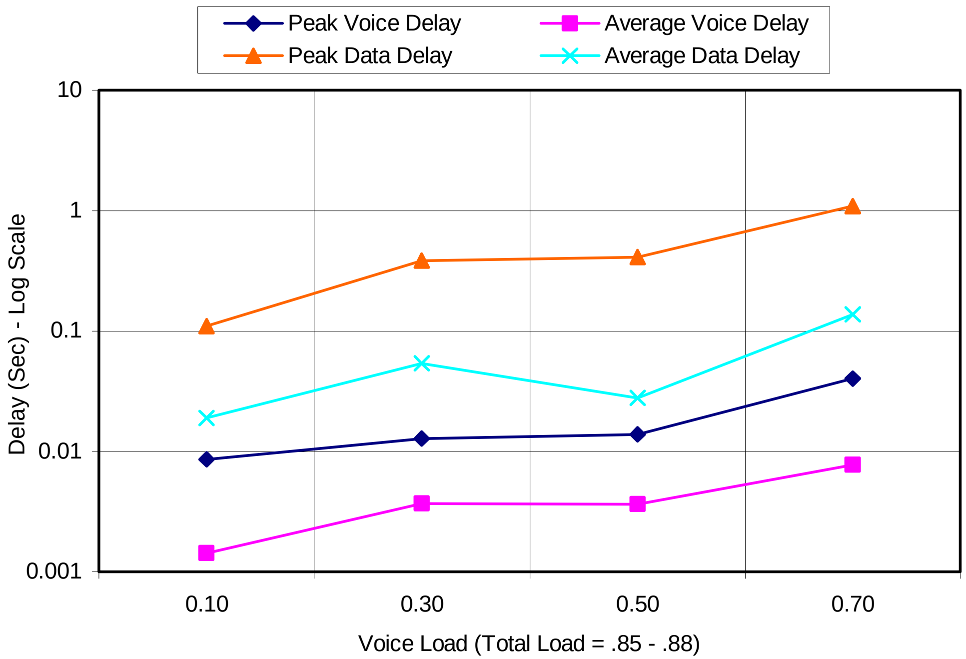



- Data load of .64, voice load of .20 - Constant
- Link speed of 288 kbps – Constant, Maximum Data MTU size 1000 bits
- Pareto shape parameter used is 1.06 and the k parameter used is 453.
- This test is a repeat of test on last slide, except that a Pareto distribution is being used for the data MTU size instead of an exponential distribution.
- Graphs show a comparison of the results with a Pareto distribution and exponential distribution

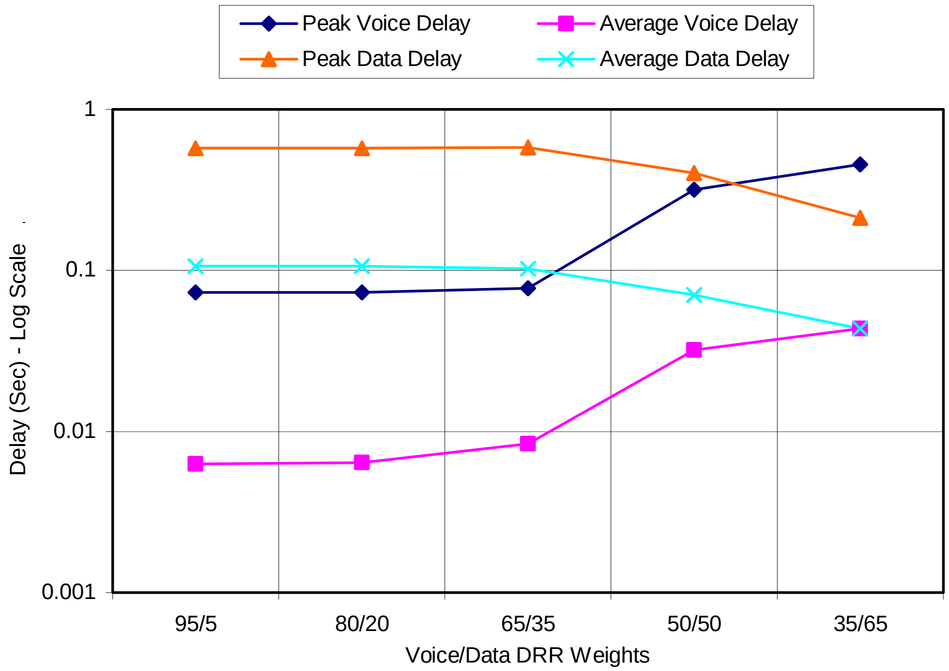
# Core Network – Load

| Voice Load | Data Load | Total Load |
|---|---|---|
| .10 | .78 | .88 |
| .30 | .57 | .87 |
| .50 | .36 | .86 |
| .70 | .16 | .86 |

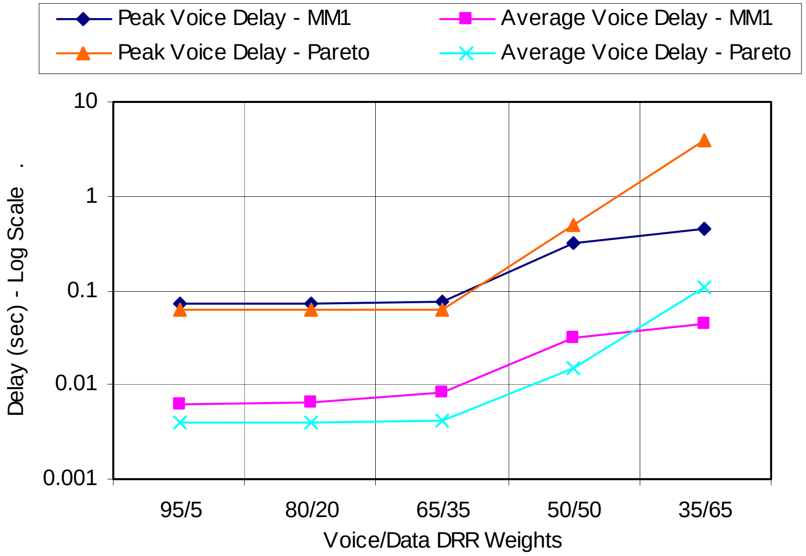


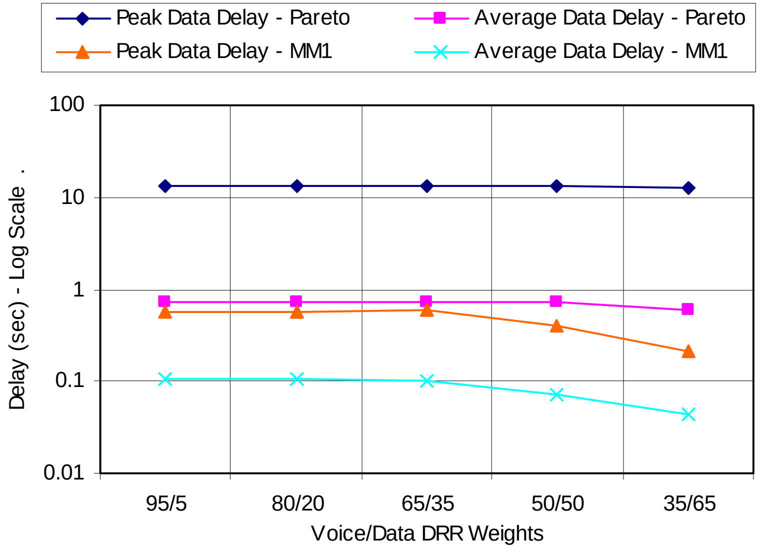
# Core Network – DRR Queue

- Voice load of .5 - Constant
- Data load of .36 - Constant
- Maximum Data MTU size 4000 bits



# Core Network – Pareto w/ DRR




- Data/Voice Load is constant, Voice Load = .5, Data Load = .36
- Maximum Data MTU size is 4000 bits
- Pareto shape parameter used is 1.06 and the k parameter used is 453.
- This test is a repeat of Core Scenario 3, except that a Pareto distribution is being used for the data MTU size instead of an exponential distribution.
- Graphs show a comparison of the results with a Pareto distribution and exponential distribution

# Latency Analysis - Observations

- The interarrival time between voice packets (20 ms for this simulation) is very important in the determination of load and the calculation of delay for both the Remote Network and Core Network models.
- The assumption of 40% "on" time for the voice generator also played a critical role in determining maximum delay.
- When DRR queues were used in the Core Network, delay were in excess of 60 ms. When priority queues were used in the same situations, delays were as low as 40 ms. This presents problems with DRR queues in large networks. This probably occurred because a DRR queue will let a data packet start once it has built enough “quantum”, even if voice packets are waiting.
- Maximum MTU size is very important in the Remote Network Model
- [CHARNY] showed that with multiple hop counts, analytical calculation of maximum possible delay can lead to very high delay times for relatively low link utilization levels. This study did not see those delay times in simulation.

# Conclusions

- The 3-Step Approach proposed to Analyze Mission Critical Voice was partially completed. The two analyses were completed (Voice Quality, Latency) and were successful.
- The Voice Quality Analysis using the E-Model Optimization worked properly and was an effective tool to help choose parameters like coder, packet loss level, and utilization in a Mission Critical Voice over IP network.
- The Latency Analysis showed that even with very tight control over the network parameters, it is difficult (but still possible) to meet the requirements for mission critical voice over IP. Strict control over load, MTU size, file size distribution is required. This may or may not be possible in different mission critical networks.

# Topics for Future Research

- Research is needed that studies the survivability and availability of mission critical voice over IP networks. More research is needed into ways to tie voice quality and latency into availability analysis.
- More accurate methods to model large IP networks are needed. This research modeled a large network, but frequently assumptions were necessary to manage the size of the model.
- More accurate estimates of delay bounds in core networks are needed.
- More research is needed to extend the optimization of the E-Model to include more variables, which will increase its usefulness. In addition, better estimates of network latency will help this model to be more accurate.

# Questions?