ENSC 427: COMMUNICATION NETWORKS PROJECT

ANALYSIS OF LONG DISTANCE 3-WAY CONFERENCE CALLING WITH VOIP

SPRING 2010

FINAL PROJECT PRESENTATION

GROUP #6:

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Outline

- Motivation
- Project Overview
- Related work
- Project description: Network Topology
- Implemented Scenarios
- Conclusion
- Questions

Motivation

- VoIP is an alternative to circuit-switched networks.
- It provides a means of communication using IP networks over local and long distances.
- Free: local and long-distance calls.
- Flexibility to make and receive calls as long as there is a broadband connection.
- Diverse features: caller ID, call waiting and call forwarding, voicemail, and three-way conference calling.
- Good voice quality, as good as cell phone voice quality.
- Increase in popularity.

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Overview

- This project will analyze the performance of long distance 3-way voice conference calling using VoIP

- Performance will be evaluated by examining:
  - Packet loss
  - End-to-end packet delay
  - Delay jitter
  - Speech quality (MOS)

Overview

- We will attempt to test the network by varying:
  - Voice codecs
  - Background and link loads

- A voice call can be declared explicitly in OPNET using the voice application traffic model. Then through testing, we can determine the call quality through subjective and quantitative means.

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Related work

- Speaker selection for tandem-free operation VoIP Conference bridges.
- Impact of link failures on VoIP performance using passive and active traffic measurements.
- Performance analysis of VoIP traces using Software and hardware phones. QoS is evaluated by observing parametrics such as packet loss, jitter, and by the client estimation using Mean Opinion Score.

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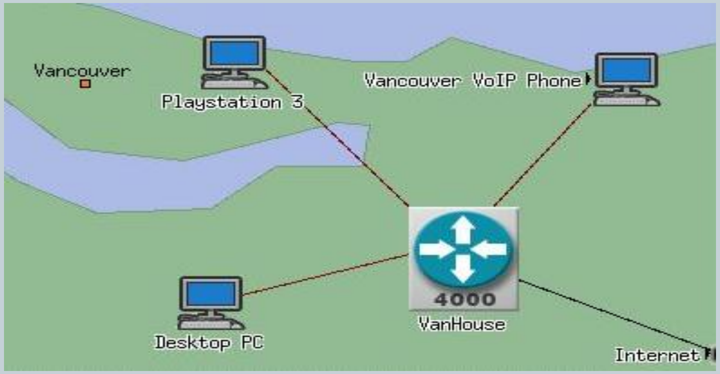
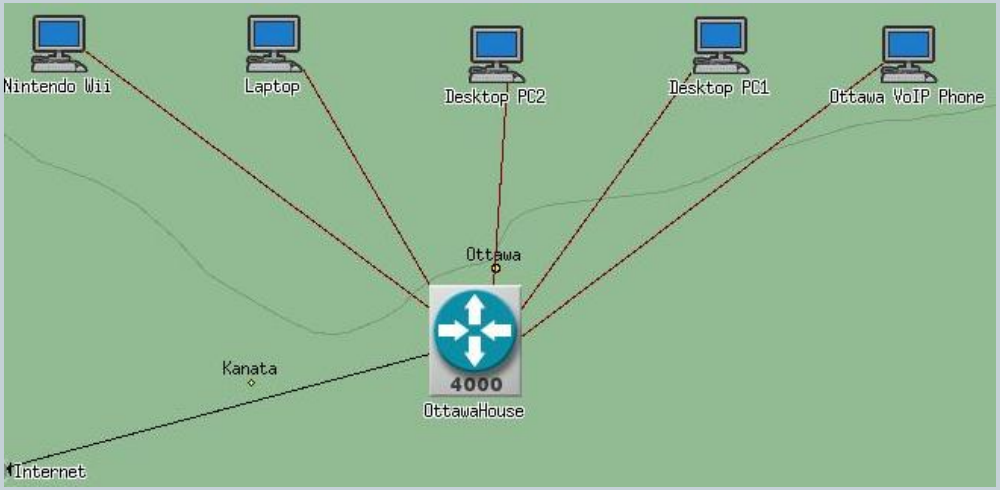
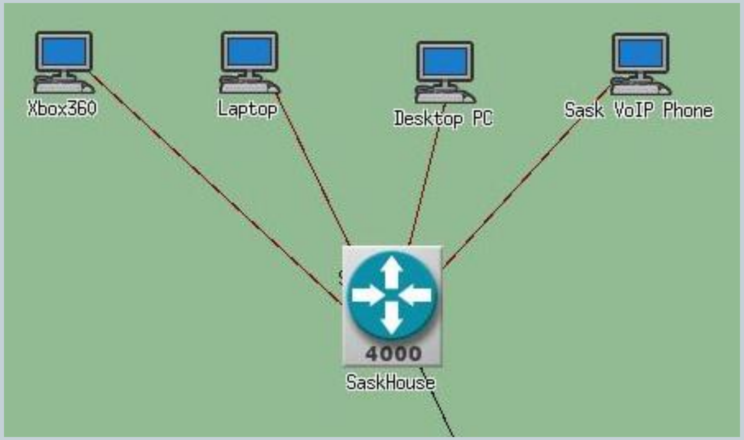
Project Description: Network Topology

Subnets: Vancouver, Saskatoon, Ottawa







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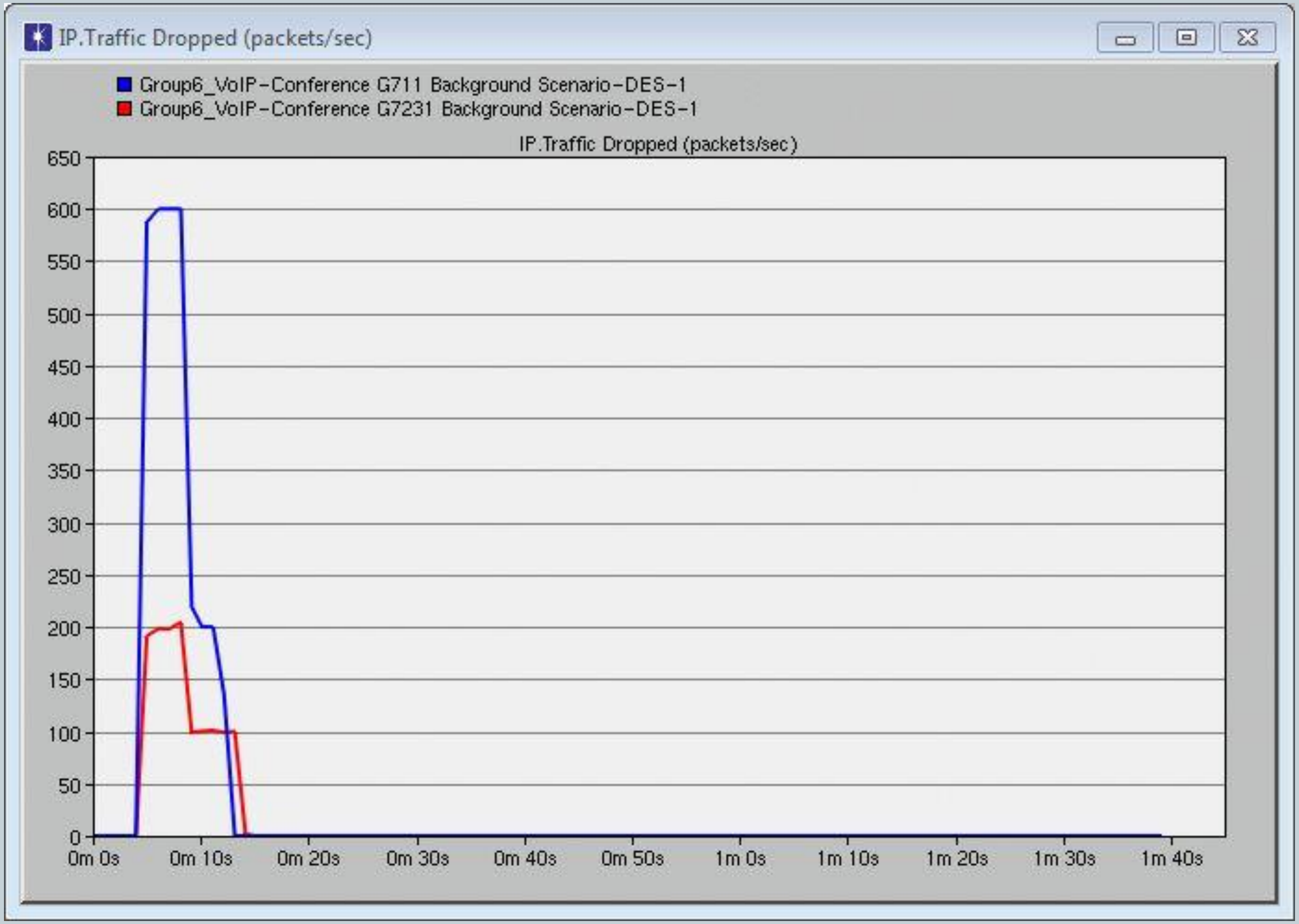
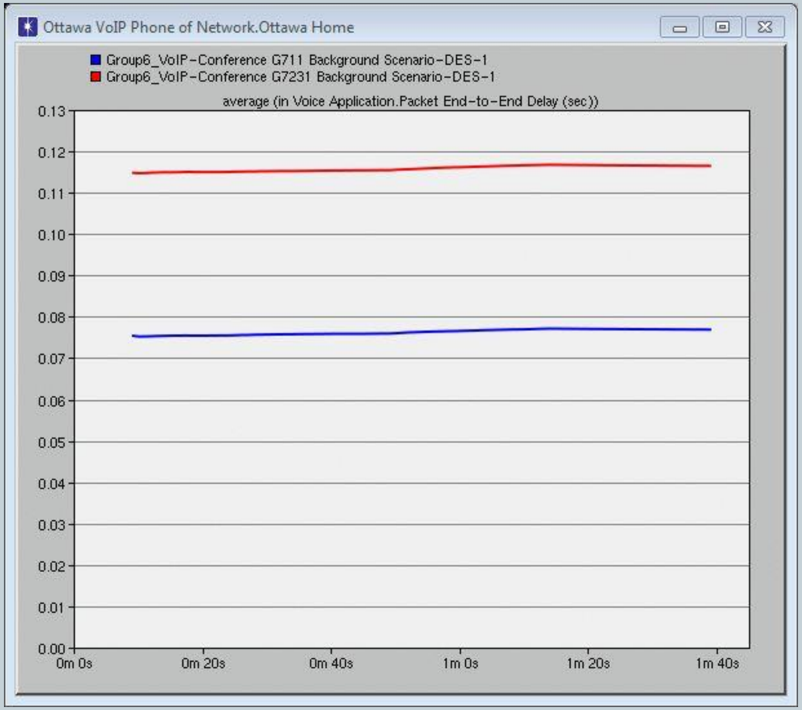
Implemented Scenarios:

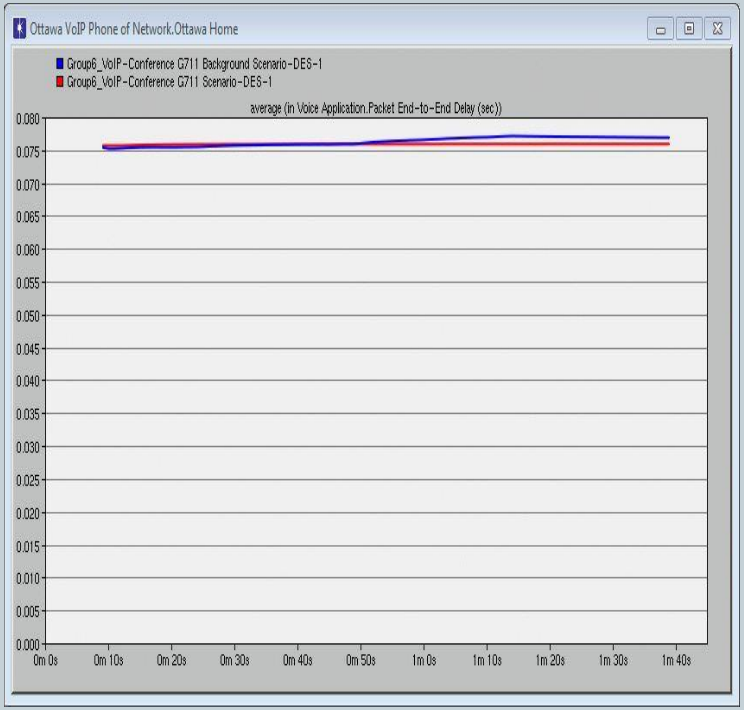
- Scenario I:
  - Conference Call with G711 Vocoder
- Scenario II:
  - Conference Call with G711 Vocoder & Background Load
- Scenario III:
  - Conference Call with G728 Wideband Vocoder
- Scenario IV:
  - Conference Call with G728 Vocoder & Background Load

Packet Loss:

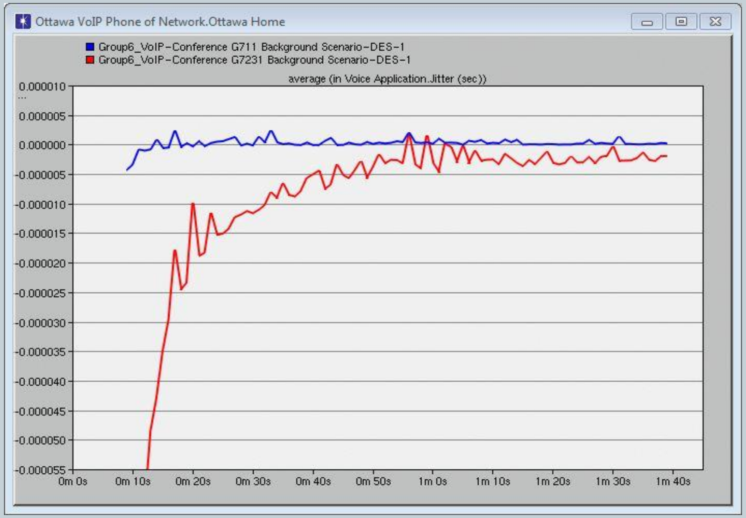

IP.Traffic Dropped (packets/sec)
Group6_VoIP-Conference G711 Background Scenario-DES-1
Group6_VoIP-Conference G7231 Background Scenario-DES-1
IP.Traffic Dropped (packets/sec)
650
600
550
500
450
400
350
300
250
200
150
100
50
0
0m 0s
0m 10s
0m 20s
0m 30s
0m 40s
0m 50s
1m 0s
1m 10s
1m 20s
1m 30s
1m 40s

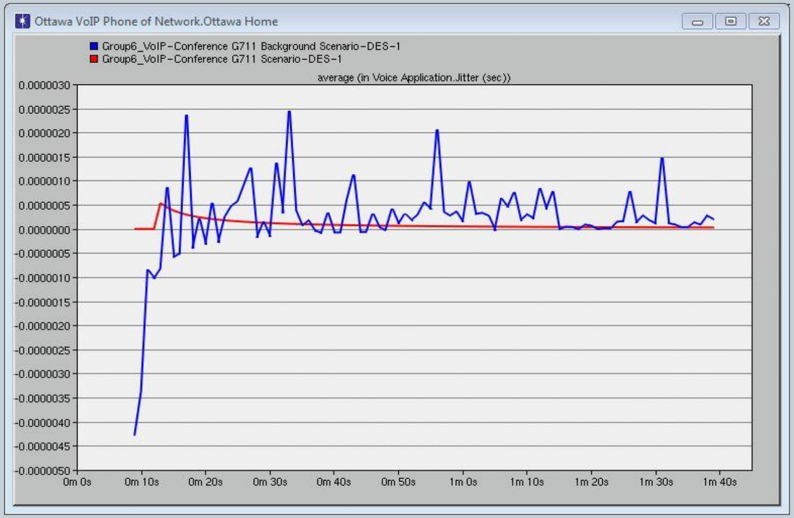
End-to-End Packet Delay:

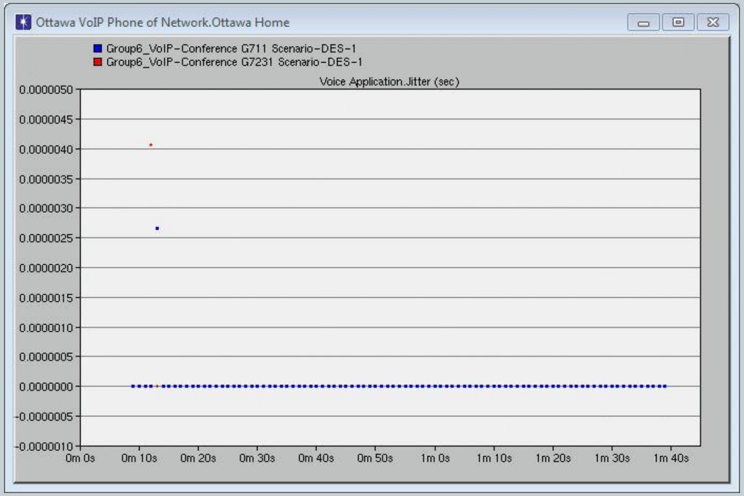


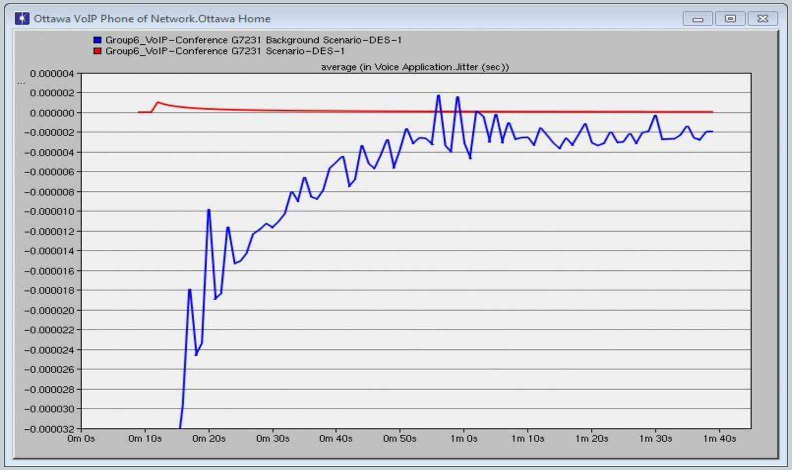


Delay Jitter:

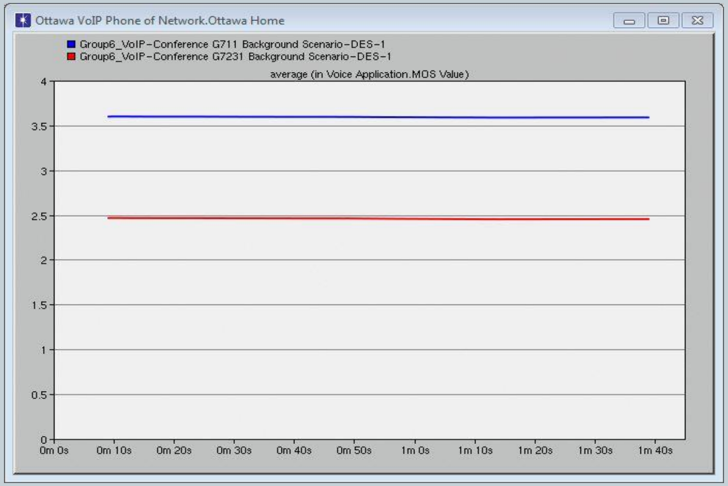


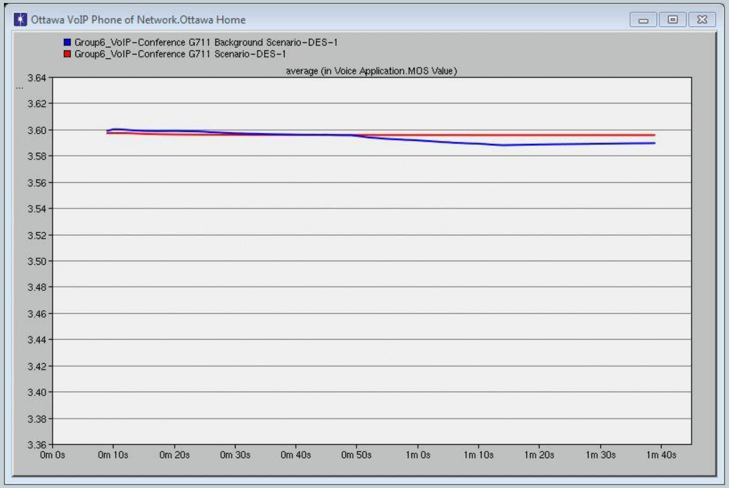


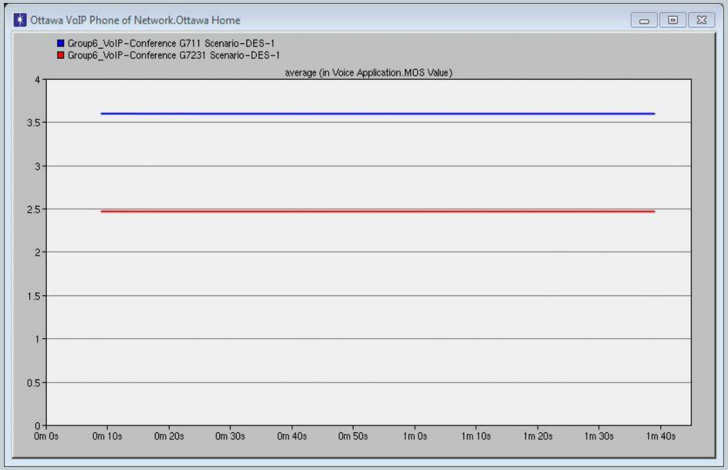


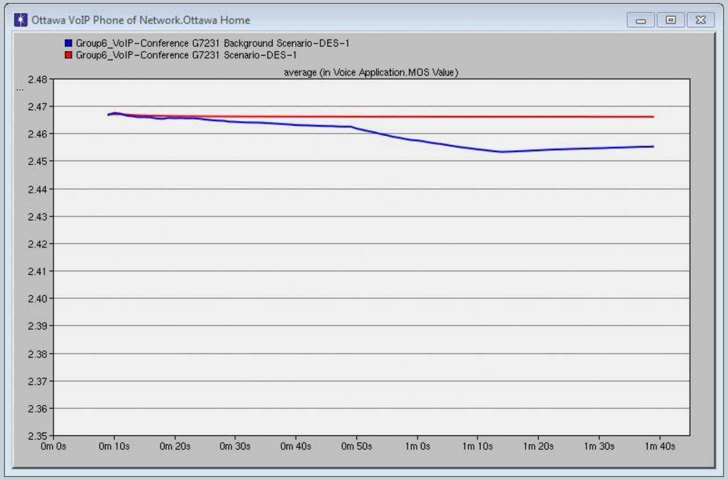


Speech Quality (MOS):









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Conclusion

- Packet Loss:
- End-to-End Packet Delay:
- Delay Jitter:
- Speech Quality (MOS):

References

[1] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, “RTP: A transport protocol for real-time applications,” IETF RFC 3550, Jul. 2003.

[2] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, “SIP: Session Initiation Protocol,” IETF RFC 3261, Jun. 2002.

[3] B. Goode, “Voice Over Internet Protocol VoIP,” Proc. IEEE, vol. 90, pp. 1495-1516, Sept. 2002.

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[5] P. Smith, P. Kabal, and R. Rabipour, “Speaker Selection For Tandem-Free Operation VoIP Conference”, Proc. IEEE Workshop Speech Coding, pp. 120-122, Oct. 2002.

[6] C. Boutremans, G. Iannaccone, and C. Diot, “Impact of link failures on VoIP performance,” Proc. NOSSDAV, May 2002.

[7] R. G. Cole and J. H. Rosenbluth, “Voice over IP performance monitoring,” Computer Communication Review, vol. 31, no. 2, pp. 9–24, April 2001.

Questions