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INTEGRATION PROTOCOLS FOR VOICE AND DATA TRAFFIC

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INTEGRATION PROTOCOLS FOR VOICE AND DATA TRAFFIC

By

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To my parents ....

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LIST OF ABBREVIATIONS

ARR	-	Adaptive Round Robin Protocol
AP	-	Absolute Priority
ATM	-	Asynchronous Transfer Mode
B-ISDN	-	Broadband Integrated Services Digital Network
BW	-	Bandwidth
CAD	-	Computer Aided Design
CBR	-	Continuous Bit Rate or Constant Bit Rate
CCITT	-	Consultative Committee for International Telegraph & Telephone
CDV	-	Cell Delay Variation
CPE	-	Customer Premises Equipment
DLSM	-	Duration Limited Statistical Multiplexing
DTS	-	Dynamic Time Slice
EDD	-	Earliest Due Date
EDF	-	Earliest Deadline First
ERR	-	Exhaustive Round Robin
FCFS	-	First Come First Served
FIFO	-	First in First out
FRP	-	Fast Reservation Protocol
FRP/DT	-	Fast Reservation Protocol with Delayed Transmission
FRR	-	Fixed Round Robin
GCRA	-	Generic Cell Rate Algorithm

GP	-	Gated Priority
GR	-	Golden Ratio
GRR	-	Gated Round Robin
HBS	-	High Bandwidth Services
HOL-PJ	-	Head of Line with Priority Jumps
IACS	-	Integrated Access and Cross-connect System
IBP	-	Interrupted Bernoulli Process
LAN	-	Local Area Network
LIFO	-	Last in First out
LQF	-	Longest Queue First
LTRB	-	Least Time to Reach Bound
MBEA	-	Most Behind Expected Arrival
MLT	-	Minimum Laxity Threshold
MMDP	-	Markov Modulated Deterministic Process
MMPP	-	Markov Modulated Poisson Process
MPQ	-	Mixed Priority Queueing
MRR	-	Modified Round Robin
NRT	-	Non Real Time
NTCD	-	Nested Threshold Cell Discarding
NTCD/MB	-	Nested Threshold Cell Discarding with Multiple Buffer
OCF	-	Oldest Customer First
PBS	-	Partial Buffer Sharing
PCM	-	Pulse Code Modulation
PJ-FT	-	Priority Jump with Fixed Threshold

PJ-MT	-	Priority Jump with Movable Threshold
PO	-	Pushout
QLT	-	Queue Length Threshold
QoS	-	Quality of Service
RR	-	Round Robin
RT	-	Real Time
SCP	-	Self Calibrating Pushout
SPS	-	Strictly Priority Servicing
SVBR	-	Step-wise Variable Bit Rate
TDM	-	Time Division Multiplexing
UNI	-	User Network Interface
VBR	-	Variable Bit Rate
VLSI	-	Very Large Scale Integration
WPT	-	Wideband Packet Technology
WRR	-	Weighted Round Robin

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INTEGRATION PROTOCOLS FOR VOICE AND DATA TRAFFIC

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Increasing demands for multimedia services offer integration of multimedia traffic as a hot issue in the future research areas. As a result, in the literature, many multiplexing schemes have been proposed. However, most of them have been implemented with a high complexity, others may be non-effective to satisfy the multiplexing performance criteria, while the rest are still not subjected to a wide range of analysis. Therefore, there is a critical need for comparing some of the recommended multiplexing schemes as well as developing a simple and effective integration protocol while still achieving reasonable bandwidth utilization.

This thesis is intended to examine integration protocols for multimedia traffic, with primary focusing on voice-data integration. Firstly, a survey of the existing multiplexing schemes and related issues are presented. Next, an Adaptive Round Robin (ARR) protocol is proposed, as an alternative for voice-data integration, and extensively simulated. Finally, further comparisons, based on computer simulations, are carried out for various multiplexing schemes including Strictly Priority Servicing (SPS), Fixed Round Robin (FRR), Dynamic Bandwidth Allocation/(T_1, T_2) and Queue Length Threshold (QLT).

As a contribution of the thesis, the proposed protocol tries to avoid the drawbacks of the previous multiplexing schemes besides satisfying the multiplexing performance criteria. The protocol differs from the others in that, it gives a limited priority for voice over data, it organizes the incoming packets to the single First-in First-out (FIFO) output buffer rather than the only outgoing scheduling, i.e., all data sources are polled in order according to the adaptation policy; however, before a data source can send a packet, all active voice sources are polled in order. Thus it provides an improvement in voice delay performance without significant effect on data delay performance over previous protocols. In addition, simulation comparisons between various multiplexing schemes have been discussed. In these simulations voice packets are assumed to be generated from on-off sources (talkspurt-silence calls), which is closer to reality and which is not considered in most of the performance analyses of previous schemes.

Abstrak tesis yang dikemukakan kepada Senat Universiti Putra Malaysia
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PROTOKOL PENYEPADUAN PADA TRAFIK SUARA DAN DATA

Oleh

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Permintaan yang kian meningkat dalam multimedia menjadikan penyepaduan multimedia sebagai suatu isu hangat untuk penyelidikan pada masa akan datang. Justeru itu, pelbagai jenis skim pemultipleksan telah dicadangkan. Walau bagaimanapun, kebanyakan daripadanya telah digunakan pada kompleksiti tinggi, yang lain pula mungkin tidak mampu untuk mencapai kriteria persembahan pemultipleksan, manakala selebihnya belum memerlukan analisis terperinci lagi. Oleh yang demikian, perbandingan beberapa skim pemultipleksan amat diperlukan dan seterusnya membina suatu protokol yang mudah dan berkesan yang boleh mencapai penggunaan jalur frekuensi yang munasabah.

Tesis ini bertujuan untuk mengkaji protokol penyepaduan pada trafik multimedia, dengan memfokuskan kepada penyepaduan suara dan data. Pertamanya, kajian tentang skim pemultipleksan yang sedia ada dan isu-isu berkaitan dibentangkan. Kemudian, Protokol Suai Bulatan Robin dicadangkan sebagai alternatif kepada penyepaduan suara-data dan disimulasikan sepenuhnya. Akhirnya, perbandingan dengan menggunakan simulasi komputer dilakukan ke atas pelbagai

jenis skim pemultipleksan termasuk Layanan Keutamaan Tegas, Bulatan Robin Tetap, Pengagihan Jalur Frekuensi Dinamik dan Ambang Panjang Baris Gilir.

Sebagai sumbangan tesis ini, protokol yang dicadangkan adalah untuk mengatasi kelemahan-kelemahan yang ada pada skim pemultipleksan yang terdahulu dan berjaya mencapai kriteria persembahan pemultipleksan yang memuaskan. Protokol ini berlainan dengan yang sedia ada iaitu ia memberi keutamaan yang terhad kepada bunyi ke atas data, ia mengaturkan paket-paket yang masuk kepada penimbal pengeluar masuk dulu keluar dulu dan bukan hanya pengeluaran berjadual sahaja, misalnya semua maklumat data akan diatur mengikut polisi penyuai. Walau bagaimanapun, sebelum sumber data menghantar suatu paket, semua maklumat suara yang aktif akan diatur dengan kemas. Oleh itu, ia memberi suatu peningkatan pada persembahan lengah-suara dengan tidak memberi sebarang kesan pada persembahan lengah-data pada protokol-protokol yang terdahulu. Tambahan lagi, perbandingan simulasi di antara pelbagai jenis skim pemultipleksan telah pun dibincang. Dalam simulasi-simulasi ini, paket suara dianggap dijanakan dari punca hidup/mati (panggilan talkspurt-senyap), yang lebih hampir dengan realiti dan ini tidak ditumpukan oleh kebanyakan analisis persembahan dalam skim-skim terdahulu.

CHAPTER I

INTRODUCTION

Future Trends in Communication

Future communication will be characterized by high bandwidth utilization, multimedia applications and both point-to-point and multipoint connections. These facts require a network with intelligence, wider bandwidth and more flexibility. New technologies and the complete standardization of new services and interfaces are necessary to achieve this objective. As a matter of fact, the increasing demand for multimedia and video services has motivated the development of the Broadband Integrated Services Digital Network (B-ISDN), a public switched telecommunications network infrastructure capable of supporting both narrowband and broadband services on a single flexible network platform and providing customer access over a single interface. Further, Asynchronous Transfer Mode (ATM) is a recommended transport technique for B-ISDN by CCITT I.121 (1988), and has been recognized as an attractive way of exchanging multimedia information such as voice, computer data, and motion video.

Introduction and evolution towards B-ISDN must, in any case, consider the strong need to interwork with existing services and networks, including telephony, 64 kbps ISDN, data (packet) networks and TV distribution networks. Another issue of similar importance is the integration of such services and networks.

Multimedia Traffic Engineering

A multimedia call may communicate audio, data, still images, or full-motion video, or any combination of these media. Each medium has different demands for communication qualities such as bandwidth requirement, signal latency within the network and signal fidelity upon delivery by the network. Moreover, the information content of each medium may affect the information generated by other media. For example, voice could be transcribed into data via voice recognition and data commands may control the way voice and video are presented. These interactions most often occur at the communication terminals but may also occur within the network.

Traffic measurements and load characteristics (peak and mean rate, burstiness) are essential for the analysis and evaluation of network performance as well as for network design. For instance, networks' researchers and designers start their work by identifying traffic parameters.

Multimedia networks must support diversity of service and performance requirements. For instance, real-time voice and video have strict delay requirements, whereas in many data applications, real-time delivery is not a primary concern. Even within delay-sensitive traffic (e.g., voice or video), different traffic streams may have different delay requirements; i.e., some may contain more urgent information than others. Some traffic is loss-sensitive (e.g., data) and thus must be received without errors, whereas the inherent structure of speech allows for some loss of information without significant quality degradation.

Due to the various quality of service (QoS) requirements for multimedia services the traffic management becomes more difficult and challenging to network designers. For example, real-time traffic requires small end-to-end delay and jitter, while data transfer requires low cell loss rates to reduce the number of retransmissions to prevent the network from being more congested.

Multiplexing of Multimedia Traffic

Each call is given a fixed amount of bandwidth in a circuit switched network. For instance, telephone network provides 64 kbps channels to end users. However, circuit switching technique can be inefficient because the transmitter for an individual call may not always need to transmit data at its allowed rate. Alternatively deterministic and statistical multiplexing are used.

With deterministic multiplexing, the sum of the peak bandwidths for the constituent connections is less than the peak bandwidth capability of the channel into which they are multiplexed. If the connections are multiplexed deterministically, then each connection is allocated sufficient resources to accommodate its peak rate. Thus congestion no longer will be an issue and decisions on the admission of new connections are based purely on the quantity of unallocated bandwidth in the network.

On the other hand, statistical multiplexing occurs when the capacity of an output channel is less than the sum of the peak connection bandwidths, but is larger than their average total bandwidth requirement. Connections are multiplexed statistically to take advantage of the variable information transfer rates generated by many applications. This approach allows many connections to implicitly share resources on the assumption that each connection only requires these resources for a small fraction of its time. The statistical gain is the factor by which the sum of the peak bandwidths exceeds the output channel's capacity. Whenever statistical multiplexing is used there is a finite probability that at some point in time the offered load to a multiplex point will exceed its output capacity. Short overloads can be accommodated by the provision of buffers. The buffers will absorb excess information until the sum of the input rates drops below the output rate of the multiplexer. The larger the buffer, the greater the overload that can be absorbed, but at the expense of delay. Buffering can help avoid but cannot eliminate congestion, as there is still a finite probability of buffer overflow, unless the duration of the overload is bound by some constraint. The straight forward solution in the case of buffer overflow is to discard information. Further, there are two different statistical multiplexing techniques that can be applied in ATM network; limited and full statistical multiplexing. The full statistical multiplexing maximizes the use of bandwidth, since all traffic types are multiplexed. However, the drawback of this technique is that it adds complexity to real time transport procedures. In limited statistical multiplexing, homogeneous services are grouped into classes and statistical multiplexing is used inside each class. Hence, it compromises between efficiency and complexity.

Multiplexing Performance Criteria

Multiplexing performance can be measured by; its impact on the packet delay variation of each connection, packet delay and buffer requirements at the multiplexer, and a fair multiplexing among the connections. Another aspect for selecting a multiplexing scheme is its feasibility to support the speed of the UNI access link without wasting outgoing slots.

Thus an effective multiplexer should make efficient use of the transmission bandwidth while still meeting the performance (e.g., delay and packet loss) requirements for all traffic types. Ideally, each traffic type should be protected from overloads caused by the other traffic in the multiplexer. Further, the multiplexer should have some congestion control schemes to allow for graceful degradation of performance when overloads occur.

Integration of Multimedia Traffic

Traditionally, communication services are carried via separate networks; voice on the telephone network, data on computer networks or local area networks, video teleconferencing on private corporate networks, and television on broadcast radio or cable networks. These networks are largely engineered for a specific application and are ill-suited for other applications.

Today integration of such services and networks is highly needed. Integration within the network can have different meanings, depending on the part and the function of the network being considered. For instance, integrated access, integrated transport, integrated switching and integrated call processing.

Integrated access involves the sharing, among services from an end user, of a single interface to a single transmission medium in the local access network. A well integrated access network should provide flexible multiplexing of as many services as possible.

Integrated transport involves the flexible sharing, among services from possibly many users, of high capacity transmission links in the network. Integrated transport avoids the segregation of different traffic types and media onto different transmission links, hence may facilitate easier interactions between media within the network.

Integrated switching involves switching multi-rate, multi-media services within a single switching machine in particular a single interconnection network. An integrated switch would avoid the necessity of adding a new interconnection network whenever a new service of distinct traffic characteristic is introduced. An integrated switching network must be flexible enough to meet the delay and bit rate requirements of each service.

Integrated call processing involves the sharing of communication software for calls of different multi-media, multi-rate and multi-point configurations. Integrated call processing provides a uniform and flexible functional description of calls, and uses a single procedure to map the resource requirements of calls onto the available physical resources in the network.

Problem Identification

As a result of multimedia services increasing demand, the integration of such services becomes a hot issue. Traditionally, voice and data are carried via separate networks, voice on the telephone network, data on computer networks or local area networks. These networks are largely engineered for a specific application and are not suitable for other applications. For instance, the traditional telephone network is too noisy and inefficient for burst data communication. On the other hand, data networks which store and forward messages using computers have very limited connectivity, usually do not have sufficient bandwidth for digitized voice and video signals and suffer from unacceptable delays for these real time signals.

Voice transmission requires a real time delivery in order to have an acceptable quality of sound. Also the voice packet loss probability should be typically less than 1%. Primary concerns of data transmission are low average delay, high throughput and good error control and recovery as packet losses could be disastrous. The requirements of computer data service are not so strict on the delay time but on the loss probability ($10^{-6} \sim 10^{-9}$), (Oh et al., 1992).

It is desirable to have a single integrated network for providing these communication services in order to achieve the economy of sharing. Integration avoids the need for many overlaying networks which complicate network management and reduce the flexibility in the introduction and evolution of services. Accordingly, in the literature many multiplexing schemes have been proposed for voice-data integration. However, most of them have been implemented with high complexity, others may not satisfy the multiplexing performance criteria, while the rest are still not subjected to a wide range of analysis. Thus there is a critical need for subjecting most of the recommended multiplexing schemes to extensive comparisons, based on computer simulations, as well as developing a simple and effective integration protocol while still achieving reasonable bandwidth utilization.

In particular, an Adaptive Round Robin (ARR) protocol is proposed as an alternative scheme for integration of voice and data. Extensive comparisons, based on computer simulations, are also examined for a group of multiplexing schemes.

Motivation

The most important research issues, which motivate the author to undertake this research, are as follows :

i- Silence periods occur naturally during a conversation (Kashorda and Jones, 1991). This is due to one person listening while the other talks. Even for the person who is talking, pauses will occur between utterances of words. These pauses will cause the link to be idle for a short period. In a properly managed link, these idle periods could be utilized for some other purpose. For instance integration of voice and data over a single channel, i.e., the network bandwidth during the silence periods could be used for data servicing.

ii- In most of the previous voice-data integration protocols, the voice traffic is usually assigned priority of service to prevent excessive delay and jitter. However, preemptive priority schemes for voice lead to degradation of data service and the requirement of excessive buffer sizes to prevent losses. Hence a limited priority to voice packets over data packets is needed.

iii- In most of the past analyses, different approaches associated with a number of traffic models have been suggested for voice-data integration. As a result, it is very difficult to differentiate between (or classify) such integration schemes. Thus performance analyses and comparisons between various multiplexing schemes, need to be further studied.

Organization of the Thesis

The work and results reported in this thesis are organized into five chapters. This chapter is an introductory chapter, in which future trends, multimedia traffic engineering, multiplexing and integration of mutlimedia traffic are highlighted. Also the research problems are identified and their motivations are introduced.