Ph.D. Thesis

Resources optimization
in multimedia communications

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Foreword

Since my M.Sc. dissertation [SR02], I have focused my research interest in the Information and Communication Technology field, with regard on the communication network resources optimization. While in [SR02] I have analysed a single network component (a multi-service switch), in this Ph.D. dissertation I have enlarged my focus to entire infrastructures based on different technologies: fixed network (using multicast services), wireless and mobile networks. Nevertheless, the thread flowing through these topics is always the same: the optimization of the resources involved. In particular I have faced the following themes:

- QoS (Quality of Service) optimization for multicast services.
- Power optimization for Wi-Fi (Wireless Fidelity) stations.
- 3G (Third Generation system) performance optimization.

In the following, a brief description of my contribution in these fields.

QoS optimization for multicast services. It focused on the group multicast routing problem: how to allocate network resources when several multicast sessions occur. The major contribution to this problem was twofold. Firstly, the use of the genetic algorithms (GAs) was proposed for solving the complexity problem inherent to the grouping of multiple sessions. Secondly, a novel cost function was proposed, weighting in a single expression both network bandwidth allocation and provided one-way delay. The proposed function was guided by few parameters that can be easily tuned during traffic engineering operations; an appropriate setting of these parameters allows the operator to configure the desired balance between network resource utilization and provided QoS (in terms of transmission delay). Experimental results were compared with the ones obtained by applying GA technique to two cost functions proposed in literature for solving this kind of problem. This highlighted that the proposed method was able to provide better solutions both in terms of link usage and end-to-end delay requirements.

Power optimization for Wi-Fi stations. In Wi-Fi infrastructured LANs (Local Area Networks) with Distributed Coordination Function (DCF), the power saving algorithm works as follows: the stations with no frames to transmit may switch to the doze mode to save power; all these stations are informed by the Access Point (AP) at the beginning of each beacon interval if there are pending frames; these stations then start contending the channel to receive these pending frames from the AP, if any. In this dissertation, a different strategy is proposed, based on giving the AP the power of deciding which stations with pending frames to notify. Indeed, there were several circumstances with high channel traffic in which it was better to defer the transmission so as to reduce the expected energy consumption. Rescheduling the notification increased the transmission latency that is on the other hand a drawback. The AP notification decision was then taken in view of the total average energy consumption and the introduced latency. Simulations had shown that the performance of the proposed method may lead to an overall energy saving of about 63% respect to the standard.

3G Performance optimization. 3G is the convergence of mobile, telephony and information systems which promises to change people's lives by enabling them to access information when, where and how they want. Different applications with heterogeneous bandwidth and QoS requirements can be handled by such a new mobile system allowing the Operator to provide a great set of services. However there are huge challenges for the Actors in the mobile telecommunications field as they rollout and deploy 3G mobile networks and services, both from technological and economical point of view. In order to successfully deploy new services in the 3G system the monitoring of network performance is vital. Accordingly, in this dissertation I outlined the objective of a performance activity for the 3G system that relied on the definition of a set of Key Performance Indicators (KPIs).

Dissertation overview

This thesis is organized as follows: Chapters I and II are intended to serve as an overview of the technologies and the problems faced, while Chapters III, IV and V describe the approaches adopted.

In particular, Chapter I provides a brief description on multicast protocol, Wi-Fi networks, UMTS (Universal Mobile Telecommunications System) network and genetic algorithms concepts, while Chapter II describes the use of GA algorithm for solving network design problem in multicast environment, the power management standard in wireless networks, and the QoS management in 3G networks.

Chapter III proposes a GA based technique for solving the multicast routing group problem, Chapter IV describes a new technique for optimizing the power management for the IEEE 802.11 wireless standard (a.k.a. Wi-Fi) and finally, the Chapter V focuses on a new procedure for performance monitoring in 3G networks.

Chapter I

Introduction

This Chapter is intended for introducing the reader into the technologies from which the research activities were started, thus providing a better comprehension for the following Chapters.

The following Sections provide brief descriptions on multicast protocol over wired networks, Wi-Fi networks, UMTS (Universal Mobile Telecommunications System) networks and Genetic Algorithms concepts.

1.1. Wired Networks: Multicast background

Traditional network computing applications involve communication between two computers. However, some important emerging applications such as LAN TV, desktop conferencing, corporate broadcasts, and collaborative computing require simultaneous communication between groups of computers. This process is known generically as multipoint communications.

There are three ways to design multipoint networking applications: unicast, broadcast, and multicast.

- *Unicast*. With a unicast design, as shown in Fig. I.1, applications can send one copy of each packet to each member of the multicast group. This technique is simple to implement, but it has significant scaling restrictions if the group is large. In addition, it requires extra bandwidth, because the same information has to be carried multiple times—even on shared links.

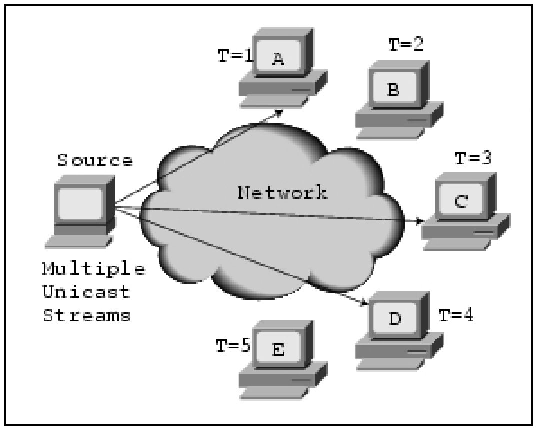

Fig. I.1. Unicast transmission

- *Broadcast*. In a broadcast design, as shown in Fig. I.2, applications can send one copy of each packet and address it to a broadcast address. This technique is even simpler than unicast for the application to implement. However, if this technique is used, the network must either stop broadcasts at the LAN boundary (a technique that is frequently used to prevent broadcast storms) or send the broadcast everywhere. Sending the broadcast everywhere is a significant usage of network resources if only a small group actually needed to see the packets. It is nearly impossible to send broadcast packets to members of a multicast group that are not within your enterprise network, such as across the Internet. Broadcast packets must be processed by each host on the network, even those not interested in the data, which places a burden on those hosts.

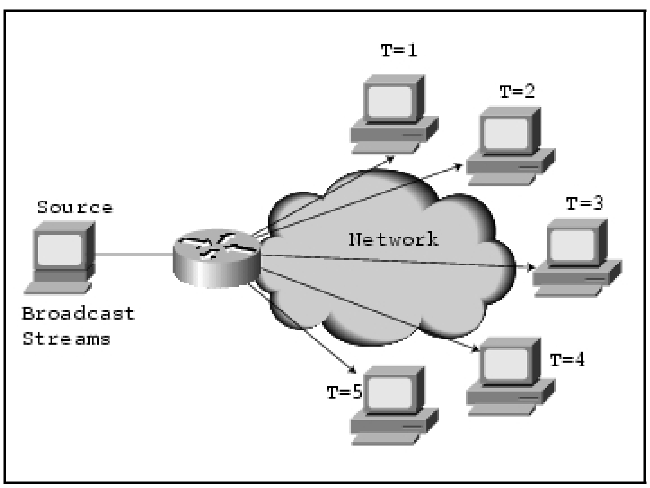

Fig. I.2. Broadcast transmission

- *Multicast.* As shown in Fig. I.3, applications send one copy of a packet and address it to a group of receivers (at the multicast address) that want to receive it rather than to a single receiver (for example, at a unicast address). Multicast depends on the network to forward the packets to only those networks and hosts that need to receive them, therefore controlling network traffic and reducing the amount of processing that hosts have to do. Multicast applications are not limited by domain boundaries but can be used throughout the entire Internet.

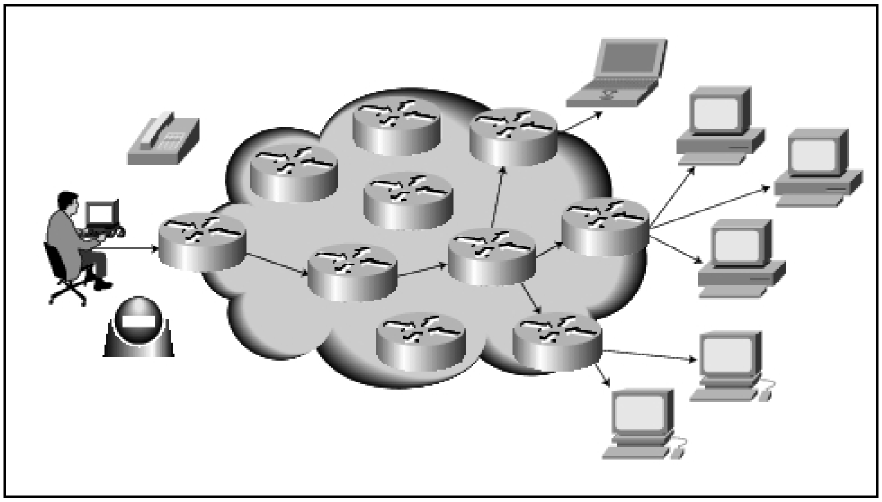

Fig. I.3. Multicast transmission

Multicast can be implemented at both the data-link layer and the network layer. Ethernet and FDDI (Fiber Distributed Data Interface), for example, support unicast, multicast, and broadcast addresses. An individual computer can listen to a unicast address, several multicast addresses, and the broadcast address. Token Ring also supports the concept of multicast addressing but uses a different technique. Token Rings have functional addresses that can be used to address groups of receivers.

If the scope of an application is limited to a single LAN, using a data-link layer multicast technique is sufficient. However, many multipoint applications are valuable precisely because they are *not* limited to a single LAN.

When a multipoint application is extended to an internet consisting of different media types, such as Ethernet, Token Ring, FDDI, ATM, Frame Relay, and other networking technologies, it is best to implement multicast at the network layer.

There are several parameters that the network layer must define in order to support multicast communications:

- *Addressing*. There must be a network-layer address that is used to communicate with a group of receivers rather than a single receiver. In addition, there must be a mechanism for mapping this address onto data-link layer multicast addresses where they exist.

- *Dynamic registration*. There must be a mechanism for the computer to communicate to the network that it is a member of a particular group. Without this ability, the network cannot know which networks need to receive traffic for each group.
- *Multicast routing*. The network must be able to build packet distribution trees that allow sources to send packets to all receivers. A primary goal of these packet distribution trees is to ensure that each packet exists only one time on any given network (that is, if there are multiple receivers on a given branch, there should only be one copy of the packets on that branch).

The Internet Engineering Task Force has been developing standards that address each of the issues described above.

- *Addressing*. The IP (Internet Protocol) address space is divided into four pieces: Class A, Class B, Class C, and Class D. Classes A, B, and C are used for unicast traffic. Class D is reserved for multicast traffic. Class D addresses are allocated dynamically.
- *Dynamic registration*. RFC (Request For Comment) 1112 defines the Internet Group Membership Protocol (IGMP). IGMP specifies how the host should inform the network that it is a member of a particular multicast group.
- *Multicast routing*. There are several standards available for routing IP Multicast traffic:
    - RFC 1075 defines the Distance Vector Multicast Routing Protocol (DVMRP).
    - RFC 1584 defines the Multicast Open Shortest Path First (MOSPF) protocol—an extension to OSPF that allows it to support IP Multicast.
    - Two Internet standards-track drafts describe PIM—a multicast protocol that can be used in conjunction with all unicast IP routing protocols. These documents are entitled *Protocol-Independent Multicast (PIM): Motivation and Architecture and Protocol-Independent Multicast (PIM): Protocol Specification*.

DVMRP (RFC 1075)

DVMRP uses a technique known as Reverse Path Forwarding. When a router receives a packet, it floods the packet out of all paths except the one that leads back to the packet's source. Doing so allows a data stream to reach all LANs (possibly multiple times). If a router is attached to a set of LANs that do not want to receive a particular multicast group, the router can send a "prune" message back up the distribution tree to stop subsequent packets from travelling where there are no members.

DVMRP will periodically reflood in order to reach any new hosts that want to receive a particular group. There is a direct relationship between the time it takes for a new receiver to get the data stream and the frequency of flooding.

DVMRP implements its own unicast routing protocol in order to determine which interface leads back to the source of the data stream. This unicast routing protocol is very like RIP (Routing Information Protocol) and is based purely on hop counts. As a result, the path that the multicast traffic follows may not be the same as the path that the unicast traffic follows.

DVMRP has significant scaling problems because of the necessity to flood frequently. This limitation is exacerbated by the fact that early implementations of DVMRP did not implement pruning.

DVMRP has been used to build the MBONE (Multicast BackBONE) —a multicast backbone across the public Internet—by building tunnels between DVMRP-capable machines. The MBONE is used widely in the research community to transmit the proceedings of various conferences and to permit desktop conferencing.

Multicast Extensions to OSPF (RFC 1584)

Multicast OSPF (MOSPF) was defined as an extension to the OSPF unicast routing protocol. OSPF works by having each router in a network understand all of the available links in the network. Each OSPF router calculates routes from itself to all possible destinations.

MOSPF works by including multicast information in OSPF link state advertisements. An MOSPF router learns which multicast groups are active on which LANs.

MOSPF builds a distribution tree for each source/group pair and computes a tree for active sources sending to the group. The tree state is cached, and trees must be recomputed when a link state change occurs or when the cache times out.

MOSPF works only in internetworks that are using OSPF. MOSPF is best suited for environments that have relatively few source/group pairs active at any given time. It will work less well in environments that have many active sources or environments that have unstable links.

PIM (Internet Draft "Protocol-Independent Multicast [PIM]:Protocol Specification")

Protocol-Independent Multicast (PIM) works with all existing unicast routing protocols. PIM supports two different types of multipoint traffic distribution patterns: dense and sparse.

Dense mode is most useful when:

- senders and receivers are in close proximity to one another;
- there are few senders and many receivers;
- the volume of multicast traffic is high;
- the stream of multicast traffic is constant.

Dense-mode PIM uses Reverse Path Forwarding and looks a lot like DVMRP. The most significant difference between DVMRP and dense-mode PIM is that PIM works with whatever unicast protocol is being used; PIM does not require any particular unicast protocol.

Sparse multicast is most useful when:

- there are few receivers in a group;
- senders and receivers are separated by WAN links;
- the type of traffic is intermittent.

Sparse-mode PIM is optimized for environments where there are many multipoint data streams. Each data stream goes to a relatively small number of the LANs in the internetwork. For these types of groups, Reverse Path Forwarding techniques waste bandwidth. Sparse-mode PIM works by defining a Rendezvous Point. When a sender wants to send data, it first sends to the Rendezvous Point. When a receiver wants to receive data, it registers with the Rendezvous Point. Once the data stream begins to flow from sender to Rendezvous Point to receiver, the routers in the path will optimize the path automatically to remove any unnecessary hops. Sparse-mode PIM assumes that no hosts want the multicast traffic unless they specifically ask for it.

PIM is able to simultaneously support dense mode for some multipoint groups and sparse mode for others.

1.2. Wireless Networks: Wi-Fi background

A Wireless LAN (WLAN or Wi-Fi) is a data transmission system designed to provide location-independent network access between computing devices by using radio waves rather than a cable infrastructure. In the corporate enterprise, wireless LANs are usually implemented as the final link between the existing wired network and a group of client computers, giving these users wireless access to the full resources and services of the corporate network across a building or campus setting.

The widespread acceptance of WLANs depends on industry standardization to ensure product compatibility and reliability among the various manufacturers. The 802.11 specification [Wi99] as a standard for wireless LANS was ratified by the Institute of Electrical and Electronics Engineers (IEEE) in the year 1997. This version of 802.11 provides for 1 Mbps and 2 Mbps data rates and a set of fundamental signalling methods and other services. Like all IEEE 802 standards, the 802.11 standards focus on the bottom two levels the ISO (International Organization for Standardization) model, the physical layer and link layer (see figure below). Any LAN application, network operating system, protocol, including TCP/IP (Transmission Control Protocol/Internet Protocol) and Novell NetWare, will run on an 802.11-compliant WLAN as easily as they run over Ethernet.

The major motivation and benefit from Wireless LANs is increased mobility. Differently from conventional network connections, network users can move about almost without restriction and access LANs from nearly anywhere.

The other advantages for WLAN include cost-effective network setup for hard-to-wire locations such as older buildings and solid-wall structures and reduced cost of ownership-particularly in dynamic environments requiring frequent modifications, thanks to minimal wiring and installation costs per device and user. WLANs liberate users from dependence on hard-wired access to the network backbone, giving them anytime, anywhere network access. This freedom to roam offers numerous user benefits for a variety of work environments, such as:

- immediate bedside access to patient information for doctors and hospital staff;
- easy, real-time network access for on-site consultants or auditors;
- improved database access for roving supervisors such as production line managers, warehouse auditors, or construction engineers;
- simplified network configuration for temporary setups such as trade shows or conference rooms;
- faster access to customer information for service vendors and retailers, resulting in better service and improved customer satisfaction;

- location-independent access for network administrators, for easier on-site troubleshooting and support;
- real-time access to study group meetings and research links for students.

Each computer, mobile, portable or fixed, is referred to as a station in 802.11. The difference between a portable and mobile station is that a portable station moves from point to point but is only used at a fixed point. Mobile stations access the LAN during movement.

When two or more stations come together to communicate with each other, they form a Basic Service Set (BSS). The minimum BSS consists of two stations. 802.11 LANs use the BSS as the standard building block.

A BSS that stands alone and is not connected to a base is called an Independent Basic Service Set (IBSS) or is referred to as an Ad-Hoc Network. An ad-hoc network is a network where stations communicate only peer to peer. There is no base and no one gives permission to talk. Mostly these networks are spontaneous and can be set up rapidly. Ad-Hoc or IBSS networks are characteristically limited both temporally and spatially.

When BSS's are interconnected the network becomes one with infrastructure. 802.11 infrastructure has several elements. Two or more BSS's are interconnected using a Distribution System or DS. This concept of DS increases network coverage. Each BSS becomes a component of an extended, larger network. Entry to the DS is accomplished with the use of Access Points (AP). An access point is a station, thus addressable. So, data moves between the BSS and the DS with the help of these access points.

Creating large and complex networks using BSS's and DS's leads us to the next level of hierarchy, the Extended Service Set or ESS. The beauty of the ESS is the entire network looks like an independent basic service set to the Logical Link Control layer (LLC). This means that stations within the ESS can communicate or even move between BSS's transparently to the LLC.

One of the requirements of IEEE 802.11 is that it can be used with existing wired networks. 802.11 solved this challenge with the use of a Portal. A portal is the logical integration between wired LANs and 802.11. It also can serve as the access point to the DS. All data going to an 802.11 LAN from an 802.X LAN must pass through a portal. It thus functions as bridge between wired and wireless.

The implementation of the DS is not specified by 802.11. Therefore, a distribution system may be created from existing or new technologies. A point-to-point bridge connecting LANs in two separate buildings could become a DS.

While the implementation for the DS is not specified, 802.11 does specify the services, which the DS must support. Services are divided into two sections:

- Station Services (SS),
- Distribution System Services (DSS).

There are five services provided by the DSS

1. Association.
2. Reassociation.
3. Disassociation.
4. Distribution.
5. Integration.

The first three services deal with station mobility. If a station is moving within its own BSS or is not moving, the stations mobility is termed No-transition. If a station moves between BSS's within the same ESS, its mobility is termed BSS-transition. If the station moves between BSS's of differing ESS's it is ESS transition. A station must affiliate itself with the BSS infrastructure if it wants to use the LAN. This is done by Associating itself with an access point. Associations are dynamic in nature because stations move, turn on or turn off. A station can only be associated with one AP. This ensures that the DS always knows where the station is.

Association supports no-transition mobility but is not enough to support BSS-transition. Enter Reassociation. This service allows the station to switch its association from one AP to another. Both association and reassociation are initiated by the station. Disassociation is when the association between the station and the AP is terminated. This can be initiated by either party. A disassociated station cannot send or receive data. ESS-transition are not supported. A station can move to a new ESS but will have to reinitiate connections.

Distribution and Integration are the remaining DSS's. Distribution is simply getting the data from the sender to the intended receiver. The message is sent to the local AP (input AP), then distributed through the DS to the AP (output AP) that the recipient is associated with. If the sender and receiver are in the same BSS, the input and out AP's are the same. So the distribution service is logically invoked whether the data is going through the DS or not. Integration is when the output AP is a portal. Thus, 802.x LANs are integrated into the 802.11 DS.

Station services are:

1. Authentication.
2. Deauthentication.
3. Privacy.
4. MAC (Medium Access Control) Service Data Unit (MSDU) Delivery.

With a wireless system, the medium is not exactly bounded as with a wired system. In order to control access to the network, stations must first establish their identity. This is much like trying to enter a radio net in the military.

Before you are acknowledged and allowed to converse, you must first pass a series of tests to ensure that you are who you say you are. That is really all authentication is. Once a station has been authenticated, it may then associate itself. The authentication relationship may be between two stations inside an IBSS or to the AP of the BSS. Authentication outside of the BSS does not take place.

There are two types of authentication services offered by 802.11. The first is Open System Authentication. This means that anyone who attempts to authenticate will receive authentication. The second type is Shared Key Authentication. In order to become authenticated the users must be in possession of a shared secret. The shared secret is implemented with the use of the Wired Equivalent Privacy (WEP) privacy algorithm. The shared secret is delivered to all stations ahead of time in some secure method (such as someone walking around and loading the secret onto each station).

Deauthentication is when either the station or AP wishes to terminate a stations authentication. When this happens the station is automatically disassociated. Privacy is an encryption algorithm, which is used so that other 802.11 users cannot eavesdrop on your LAN traffic. IEEE 802.11 specifies Wired Equivalent Privacy (WEP) as an optional algorithm to satisfy privacy. If WEP is not used then stations are “in the clear” or “in the red”, meaning that their traffic is not encrypted. Data transmitted in the clear are called plaintext. Data transmissions, which are encrypted, are called ciphertext. All stations start “in the red” until they are authenticated. MSDU delivery ensures that the information in the MAC service data unit is delivered between the medium access control service access points.

The bottom line is this, authentication is basically a network wide password. Privacy is whether or not encryption is used. Wired Equivalent Privacy is used to protect authorized stations from eavesdroppers. WEP is reasonably strong. The algorithm can be broken in time. The relationship between breaking the algorithm is directly related to the length of time that a key is in use. So, WEP allows for changing of the key to prevent brute force attack of the algorithm. WEP can be implemented in hardware or in software. One reason that WEP is optional is because encryption may not be exported from the United States. This allows 802.11 to be a standard outside the U.S. albeit without the encryption.

IEEE 802.11 Physical Layer

The three physical layers originally defined in 802.11 included two spread-spectrum radio techniques and a diffuse infrared specification. The radio-based standards operate within the 2.4 GHz ISM (Industrial, Scientific, Medical) band. These frequency bands are recognized by international regulatory agencies radio operations. As such, 802.11-based products do not require user licensing or special training. Spread-spectrum techniques, in addition to satisfying regulatory requirements, increase reliability, boost throughput, and allow many unrelated products to share the spectrum without explicit cooperation and with minimal interference.

The original 802.11 wireless standard defines data rates of 1 Mbps and 2 Mbps via radio waves using frequency hopping spread spectrum (FHSS) or direct sequence spread spectrum (DSSS). It is important to note that FHSS and DSSS are fundamentally different signalling mechanisms and will not interoperate with one another.

Using the frequency hopping technique, the 2.4 GHz band is divided into 75 1-MHz subchannels. The sender and receiver agree on a hopping pattern, and data is sent over a sequence of the subchannels. Each conversation within the 802.11 network occurs over a different hopping pattern, and the patterns are designed to minimize the chance of two senders using the same subchannel simultaneously.

FHSS techniques allow for a relatively simple radio design, but are limited to speeds of no higher than 2 Mbps. This limitation is driven primarily by FCC (Federal Communications Commission USA) regulations that restrict subchannel bandwidth to 1 MHz. These regulations force FHSS systems to spread their usage across the entire 2.4 GHz band, meaning they must hop often, which leads to a high amount of hopping overhead.

In contrast, the direct sequence signalling technique divides the 2.4 GHz band into 14 22-MHz channels. Adjacent channels overlap one another partially, with three of the 14 being completely non-overlapping. Data is sent across one of these 22 MHz channels without hopping to other channels.

To compensate for noise on a given channel, a technique called “chipping” is used. Each bit of user data is converted into a series of redundant bit patterns called “chips”. The inherent redundancy of each chip combined with spreading the signal across the 22 MHz channel provides for a form of error checking and correction; even if part of the signal is damaged, it can still be recovered in many cases, minimizing the need for retransmissions.

IEEE 802.11 Data Link Layer

The data link layer within 802.11 consists of two sublayers: Logical Link Control (LLC) and Media Access Control (MAC).

802.11 uses the same 802.2 LLC and 48-bit addressing as other 802 LANs, allowing for very simple bridging from wireless to IEEE wired networks, but the MAC is unique to WLANs.

The 802.11 MAC is very similar in concept to 802.3, in that it is designed to support multiple users on a shared medium by having the sender sense the medium before accessing it.

For 802.3 Ethernet LANs, the Carrier Sense Multiple Access with Collision Detection (CSMA/CD) protocol regulates how Ethernet stations establish access to the wire and how they detect and handle collisions that occur when two or more devices try to simultaneously communicate over the LAN. In an 802.11 WLAN, collision detection is not possible due to what is known as the "near/far" problem: to detect a collision, a station must be able to transmit and listen at the same time, but in radio systems the transmission drowns out the ability of the station to "hear" a collision.

To account for this difference, 802.11 uses a slightly modified protocol known as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) or the Distributed Coordination Function (DCF). CSMA/CA attempts to avoid collisions by using explicit packet acknowledgment (ACK), which means an ACK packet is sent by the receiving station to confirm that the data packet arrived intact.

CSMA/CA works as follows. A station wishing to transmit senses the air, and, if no activity is detected, the station waits an additional, randomly selected period of time and then transmits if the medium is still free. If the packet is received intact, the receiving station issues an ACK frame that, once successfully received by the sender, completes the process. If the ACK frame is not detected by the sending station, either because the original data packet was not received intact or the ACK was not received intact, a collision is assumed to have occurred and the data packet is transmitted again after waiting another random amount of time.

CSMA/CA thus provides a way of sharing access over the air. This explicit ACK mechanism also handles interference and other radio related problems very effectively. However, it does add some overhead to 802.11 that 802.3 does not have, so that an 802.11 LAN will always have slower performance than an equivalent Ethernet LAN.

Another MAC-layer problem specific to wireless is the "hidden node" issue, in which two stations on opposite sides of an access point can both "hear" activity from an access point, but not from each other, usually due to distance or an obstruction.

To solve this problem, 802.11 specifies an optional Request to Send/Clear to Send (RTS/CTS) protocol at the MAC layer. When this feature is in use, a sending station transmits an RTS and waits for the access point to reply with a CTS. Since all stations in the network can hear the access point, the CTS causes them to delay any intended transmissions, allowing the sending station to transmit and receive a packet acknowledgment without any chance of collision. The Fig. I.4 shows this so called four-way handshake.

In Fig. I.4, 'node A' has a frame to send; it initiates the process by sending an RTS frame. The RTS frame serves several purposes: in addition to reserving the radio link for transmission, it silences any stations that hear it. If the target station receives an RTS, it responds with a CTS. Like the RTS frame, the CTS frame silences stations in the immediate vicinity. Once the RTS/CTS exchange is complete, 'node A' can transmit its frames without worry of interference from any hidden nodes. Hidden nodes beyond the range of the sending station are silenced by the CTS from the receiver. When the RTS/CTS clearing procedure is used, any frames must be positively acknowledged. It has to be reminded that 'A' or 'B', in a infrastructure network, has to be the Access Point.

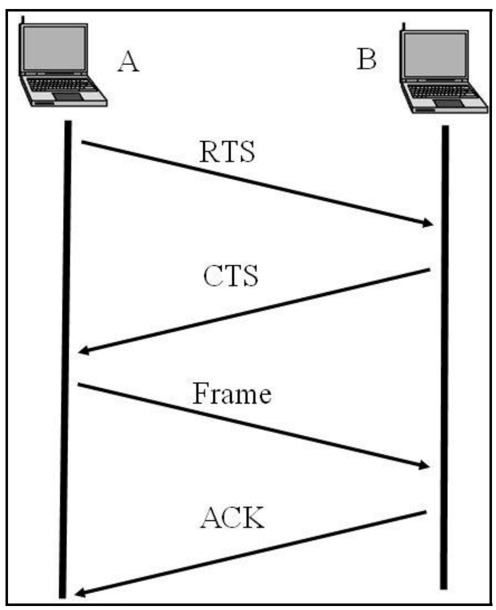

Fig. I.4. RTS/CTS clearing

Since RTS/CTS adds additional overhead to the network by temporarily reserving the medium, it is typically used only on the largest-sized packets, for which retransmission would be expensive from a bandwidth standpoint.

Finally, the 802.11 MAC layer provides for two other robustness features: CRC (Cyclic Redundancy Checking) checksum and packet fragmentation. Each packet has a CRC checksum calculated and attached to ensure that the data was not corrupted in transit. This is different from Ethernet, where higher-level protocols such as TCP handle error checking. Packet fragmentation allows large packets to be broken into smaller units when sent over the air, which is useful in very congested environments or when interference is a factor, since larger packets have a better chance of being corrupted. This technique reduces the need for retransmission in many cases and thus improves overall wireless network performance. The MAC layer is responsible for reassembling fragments received, rendering the process transparent to higher level protocols.

Time-bounded data such as voice and video is supported in the 802.11 MAC specification through the Point Coordination Function (PCF). As opposed to the DCF, where control is distributed to all stations, in PCF mode a single access point controls access to the media. If a BSS is set up with PCF enabled, time is spliced between the system being in PCF mode and in DCF (CSMA/CA) mode. During the periods when the system is in PCF mode, the access point will poll each station for data, and after a given time move on to the next station. No station is allowed to transmit unless it is polled, and stations receive data from the access point only when they are polled. Since PCF gives every station a turn to transmit in a predetermined fashion, a maximum latency is guaranteed. A downside to PCF is that it is not particularly scalable, in that a single point needs to have control of media access and must poll all stations, which can be ineffective in large networks.

Carrier-Sensing Functions and the Network Allocation Vector

Carrier sensing is used to determine if the medium is available. Two types of carriers-sensing functions in 802.11 manage this process: the physical carrier-sensing and virtual carrier-sensing functions. If either carrier-sensing function indicates that the medium is busy, the MAC reports this to higher layers.

Physical carrier-sensing functions are provided by the physical layer in question and depend on the medium and modulation used. It is difficult (or, more to the point, expensive) to build physical carrier-sensing hardware for RF-based media, because transceivers can transmit and receive simultaneously only if they incorporate expensive electronics. Furthermore, with hidden nodes potentially lurking everywhere, physical carrier-sensing cannot provide all the necessary information.

Virtual carrier-sensing is provided by the Network Allocation Vector (NAV). Most 802.11 frames carry a duration field, which can be used to reserve the medium for a fixed time period. The NAV is a timer that indicates the amount of time the medium will be reserved. Stations set the NAV to the time for which they expect to use the medium, including any frames necessary to complete the current operation. Other stations count down from the NAV to 0. When the NAV is nonzero, the virtual carrier-sensing function indicates that the medium is busy; when the NAV reaches 0, the virtual carrier-sensing function indicates that the medium is idle.

By using the NAV, stations can ensure that atomic operations are not interrupted. For example, the RTS/CTS sequence in Fig. I.4 is atomic. Fig. I.5 shows how the NAV protects the sequence from interruption. Activity on the medium by stations is represented by the shaded bars, and each bar is labelled with the frame type. Interframe spacing, further described, is depicted by the lack of any activity. Finally, the NAV timer is represented by the bars on the NAV line at the bottom of the figure. The NAV is carried in the frame headers on the RTS and CTS frames; it is depicted on its own line to show how the NAV relates to actual transmissions in the air. When a NAV bar is present on the NAV line, stations should defer access to the medium because the virtual carrier-sensing mechanism will indicate a busy medium.

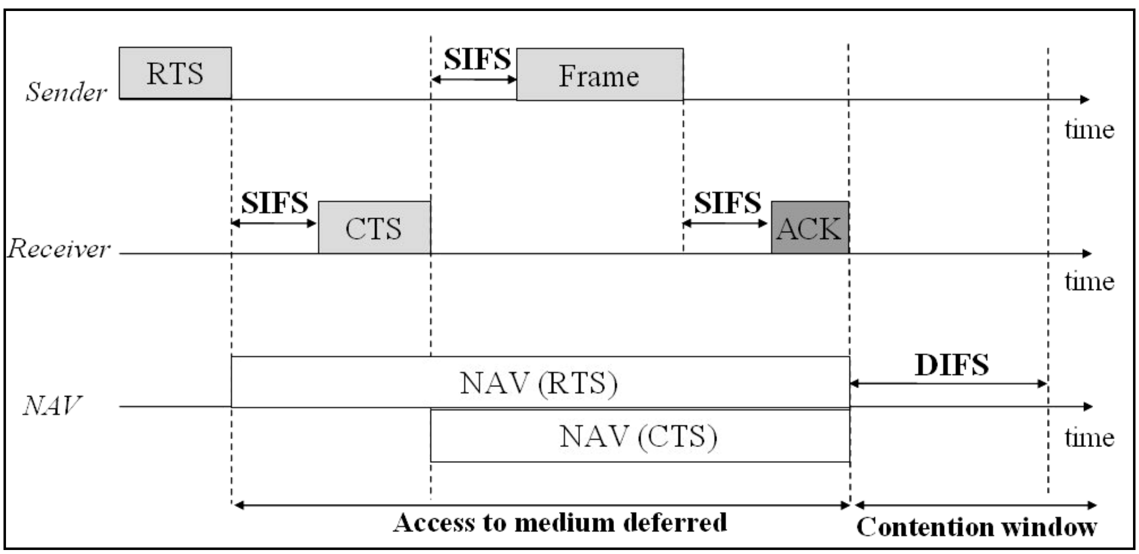

Fig. I.5. Using the NAV for virtual carrier sensing

To ensure that the sequence is not interrupted, the sender sets the NAV in its RTS to block access to the medium while the RTS is being transmitted. All stations that hear the RTS defer access to the medium until the NAV elapses.

RTS frames are not necessarily heard by every station in the network. Therefore, the recipient of the intended transmission responds with a CTS that includes a shorter NAV. This NAV prevents other stations from accessing the medium until the transmission completes. After the sequence completes, the medium can be used by any station after distributed interframe space (DIFS), which is depicted by the contention window beginning at the right side of the figure. RTS/CTS exchanges may be useful in crowded areas with multiple overlapping networks. Every station on the same physical channel receives the NAV and defers access appropriately, even if the stations are configured to be on different networks.

Interframe Spacing

As with traditional Ethernet, the interframe spacing plays a large role in coordinating access to the transmission medium. 802.11 uses four different interframe spaces. Three are used to determine medium access; the relationship between them is shown in Fig. I.6.

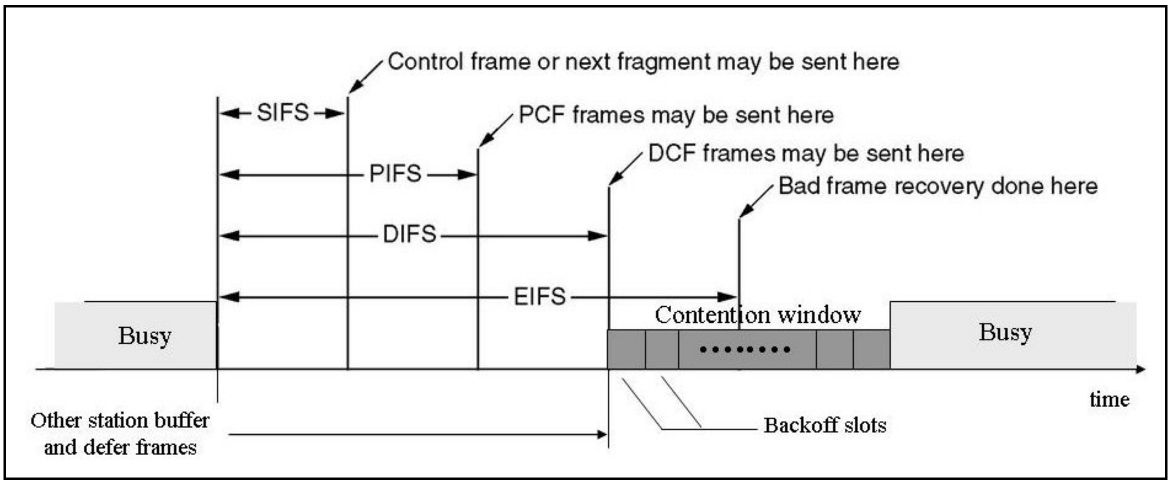

Fig. I.6. Interframe spacing relationships

It was already described that as part of the collision avoidance built into the 802.11 MAC, stations delay transmission until the medium becomes idle. Varying interframe spacings create different priority levels for different types of traffic. The logic behind this is simple: high-priority traffic doesn't have to wait as long after the medium has become idle. Therefore, if there is any high-priority traffic waiting, it grabs the network before low-priority frames have a chance to try. To assist with interoperability between different data rates, the interframe space is a fixed amount of time, independent of the transmission speed. (This is only one of the many problems caused by having different physical layers use the same radio resources, which are different modulation techniques.) Different physical layers, however, can specify different interframe space times.

- **Short interframe space (SIFS)**. The SIFS is used for the highest-priority transmissions, such as RTS/CTS frames and positive acknowledgments. High-priority transmissions can begin once the SIFS has elapsed. Once these high-priority transmissions begin, the medium becomes busy, so frames transmitted after the SIFS has elapsed have priority over frames that can be transmitted only after longer intervals.

- **PCF interframe space (PIFS)**. The PIFS, sometimes erroneously called the priority interframe space, is used by the PCF during contention-free operation. Stations with data to transmit in the contention-free period can transmit after the PIFS has elapsed and preempt any contention-based traffic.
- **DCF interframe space (DIFS)**. The DIFS is the minimum medium idle time for contention-based services. Stations may have immediate access to the medium if it has been free for a period longer than the DIFS.
- **Extended interframe space (EIFS)**. The EIFS is not a fixed interval. It is used only when there is an error in frame transmission

Atomic operations start like regular transmissions: they must wait for the DIFS before they can begin. However, the second and any subsequent steps in an atomic operation take place using the SIFS, rather than during the DIFS. This means that the second (and subsequent) parts of an atomic operation will grab the medium before another type of frame can be transmitted. By using the SIFS and the NAV, stations can seize the medium for as long as necessary.

In Fig. I.5, for example, the short interframe space is used between the different units of the atomic exchange. After the sender gains access to the medium, the receiver replies with a CTS after the SIFS. Any stations that might attempt to access the medium at the conclusion of the RTS would wait for one DIFS interval. Partway through the DIFS interval, though, the SIFS interval elapses, and the CTS is transmitted.

Contention-Based Access Using the DCF

Most traffic uses the DCF, which provides a standard Ethernet-like contention-based service. The DCF allows multiple independent stations to interact without central control, and thus may be used in either IBSS networks or in infrastructure networks.

Before attempting to transmit, each station checks whether the medium is idle. If the medium is not idle, stations defer to each other and employ an orderly exponential backoff algorithm to avoid collisions.

In distilling the 802.11 MAC rules, there is a basic set of rules that are always used, and additional rules may be applied depending on the circumstances. Two basic rules apply to all transmissions using the DCF:

1. If the medium has been idle for longer than the DIFS, transmission can begin immediately. Carrier sensing is performed using both a physical mediumdependent method and the virtual (NAV) method.

   a. If the previous frame was received without errors, the medium must be free for at least the DIFS.
   b. If the previous transmission contained errors, the medium must be free for the amount of the EIFS.
2. If the medium is busy, the station must wait for the channel to become idle. 802.11 refers to the wait as access deferral. If access is deferred, the station waits for the medium to become idle for the DIFS and prepares for the exponential backoff procedure.

Error detection and correction is up to the station that begins an atomic frame exchange. When an error is detected, the station with data must resend the frame. Errors must be detected by the sending station. In some cases, the sender can infer frame loss by the lack of a positive acknowledgment from the receiver. Retry counters are incremented when frames are retransmitted.

Each frame or fragment has a single retry counter associated with it. Stations have two retry counters: the *short retry count* and the *long retry count*. Frames that are shorter than the RTS threshold are considered to be short; frames longer than the threshold are long. Depending on the length of the frame, it is associated with either a short or long retry counter. Frame retry counts begin at 0 and are incremented when a frame transmission fails.

The short retry count is reset to 0 when:

- a CTS frame is received in response to a transmitted RTS;
- a MAC-layer acknowledgment is received after a nonfragmented transmission;
- a broadcast or multicast frame is received.

The long retry count is reset to 0 when:

- a MAC-layer acknowledgment is received for a frame longer than the RTS threshold;
- a broadcast or multicast frame is received.

In addition to the associated retry count, fragments are given a maximum “lifetime” by the MAC. When the first fragment is transmitted, the lifetime counter is started. When the lifetime limit is reached, the frame is discarded and no attempt is made to transmit any remaining fragments.

Like most other network protocols, 802.11 provides reliability through retransmission. Data transmission happens within the confines of an atomic sequence, and the entire sequence must complete for a transmission to be successful. When a station transmits a frame, it must receive an acknowledgment from the receiver or it will consider the transmission to have failed. Failed transmissions increment the retry counter associated with the frame (or fragment). If the retry limit is reached, the frame is discarded, and its loss is reported to higher-layer protocols.

One of the reasons for having short frames and long frames is to allow network administrators to customize the robustness of the network for different frame lengths. Large frames require more buffer space, so one potential application of having two separate retry limits is to decrease the long retry limit to decrease the amount of buffer space required

After frame transmission has completed and the DIFS has elapsed, stations may attempt to transmit congestion-based data. A period called the contention window or backoff window follows the DIFS. This window is divided into slots. Slot length is mediumdependent; higher-speed physical layers use shorter slot times. Stations pick a random slot and wait for that slot before attempting to access the medium; all slots are equally likely selections. When several stations are attempting to transmit, the station that picks the first slot (the station with the lowest random number) wins.

As in Ethernet, the backoff time is selected from a larger range each time a transmission fails. Fig. I.7 illustrates the growth of the contention window as the number of transmissions increases, using the numbers from the direct-sequence spread-spectrum (DSSS) physical layer. Other physical layers use different sizes, but the principle is identical. Contention window sizes are always 1 less than a power of 2 (e.g., 31, 63, 127, 255). Each time the retry counter increases, the contention window moves to the next greatest power of two. The size of the contention window is limited by the physical layer. For example, the DS physical layer limits the contention window to 1023 transmission slots.

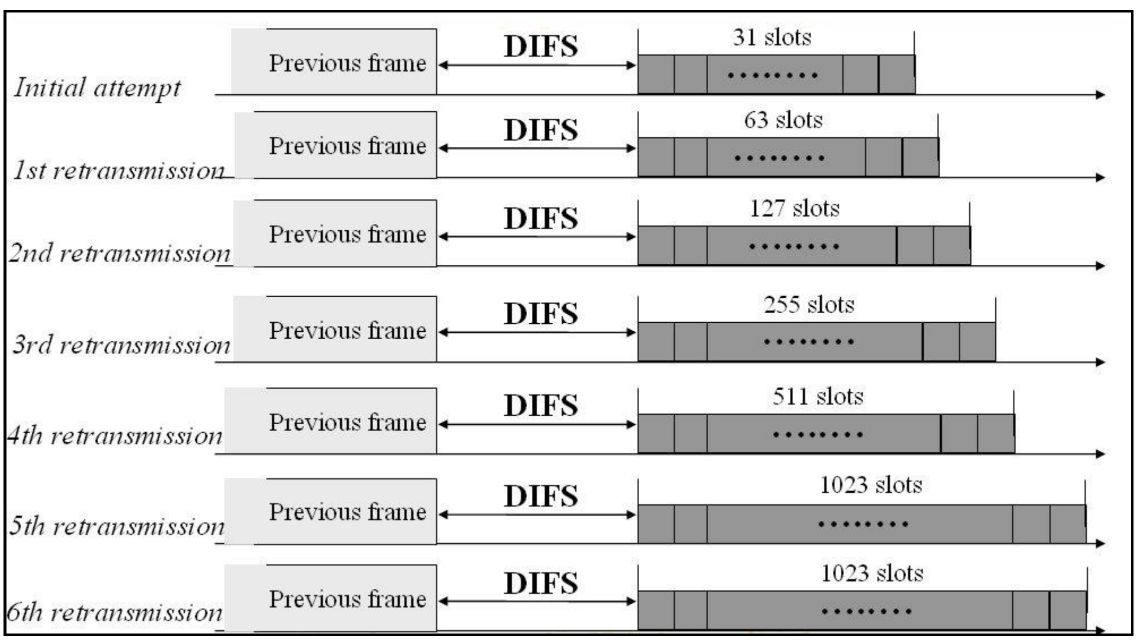

Fig. I.7. DSSS contention window size

When the contention window reaches its maximum size, it remains there until it can be reset. Allowing long contention windows when several competing stations are attempting to gain access to the medium keeps the MAC algorithms stable even under maximum load. The contention window is reset to its minimum size when frames are transmitted successfully, or the associated retry counter is reached, and the frame is discarded.

IEEE 802.11 Standards

The most critical issue affecting WLAN demand has been limited throughput. The data rates supported by the original 802.11 standards are too slow to support most general business requirements and slowed the adoption of WLANs.

Recognizing the critical need to support higher data-transmission rates, the IEEE ratified the 802.11b standard (also known as 802.11 High Rate) for transmissions of up to 11 Mbps.

After 802.11b one more standard 802.11a has been ratified and in January 2002 the draft specification of another 802.11g has been approved and ratified on early 2003.

The letters after the number “802.11” indicates the order in which the standards were first proposed. This means that the “new” 802.11a is actually older than the currently used 802.11b, which just happened to be ready first because it was based on relatively simple technology-Direct Sequence Spread Spectrum (DSSS), as opposed to 802.11a’s Orthogonal Frequency Division Multiplexing (OFDM). The more complex technology provides a higher data rate: 802.11b can reach 11Mbits/sec, while 802.11a can reach 54Mbits/sec.

IEEE 802.11b

With 802.11b WLANs, mobile users can get Ethernet levels of performance, throughput, and availability.

The basic architecture, features, and services of 802.11b are defined by the original 802.11 standard. The 802.11b specification affects only the physical layer, adding higher data rates and more robust connectivity.

The key contribution of the 802.11b addition to the wireless LAN standard was to standardize the physical layer support of two new speeds,5.5 Mbps and 11 Mbps.

To accomplish this, DSSS had to be selected as the sole physical layer technique for the standard since, as frequency hopping cannot support the higher speeds without violating current FCC regulations. The implication is that 802.11b systems will interoperate with 1 Mbps and 2 Mbps 802.11 DSSS systems, but will not work with 1 Mbps and 2 Mbps 802.11 FHSS systems.

The original 802.11 DSSS standard specifies an 11-bit chipping, called a Barker sequence, to encode all data sent over the air. Each 11-chip sequence represents a single data bit (1 or 0), and is converted to a waveform, called a symbol, that can be sent over the air.

These symbols are transmitted at a 1 MSps (1 million symbols per second) symbol rate using technique called Binary Phase Shift Keying BPSK). In the case of 2 Mbps, a more sophisticated implementation called Quadrature Phase Shift Keying (QPSK) is used; it doubles the data rate available in BPSK, via improved efficiency in the use of the radio bandwidth. To increase the data rate in the 802.11b standard, advanced coding techniques are employed.

Rather than the two 11-bit Barker sequences, 802.11b specifies Complementary Code Keying (CCK), which consists of a set of 64 8-bit code words. As a set, these code words have unique mathematical properties that allow them to be correctly distinguished from one another by a receiver even in the presence of substantial noise and multipath interference (e.g., interference caused by receiving multiple radio reflections within a building).

The 5.5 Mbps rate uses CCK to encode 4 bits per carrier, while the 11 Mbps rate encodes 8 bits per carrier. Both speeds use QPSK as the modulation technique and signal at 1.375 MSps. This is how the higher data rates are obtained. To support very noisy environments as well as extended range, 802.11b WLANs use dynamic rate shifting, allowing data rates to be automatically adjusted to compensate for the changing nature of the radio channel. Ideally, users connect at the full 11 Mbps rate.

However when devices move beyond the optimal range for 11 Mbps operation, or if substantial interference is present, 802.11b devices will transmit at lower speeds, falling back to 5.5, 2, and 1 Mbps. Likewise, if the device moves back within the range of a higher-speed transmission, the connection will automatically speed up again. Rate shifting is a physical layer mechanism transparent to the user and the upper layers of the protocol stack.

One of the more significant disadvantages of 802.11b is that the frequency band is crowded, and subject to interference from other networking technologies, microwave ovens, 2.4GHz cordless phones (a huge market), and Bluetooth. There are drawbacks to 802.11b, including lack of interoperability with voice devices, and no QoS provisions for multimedia content. Interference and other limitations aside, 802.11b is the clear leader in business and institutional wireless networking and is gaining share for home applications as well.

IEEE 802.11a

802.11a, is much faster than 802.11b, with a 54Mbps maximum data rate operates in the 5GHz frequency range and allows eight simultaneous channels

802.11a uses Orthogonal Frequency Division Multiplexing (OFDM), a new encoding scheme that offers benefits over spread spectrum in channel availability and data rate.

Channel availability is significant because the more independent channels that are available, the more scalable the wireless network becomes. 802.11a uses OFDM to define a total of 8 non-overlapping 20 MHz channels across the 2 lower bands. By comparison, 802.11b uses 3 non-overlapping channels.

All wireless LANs use unlicensed spectrum; therefore they're prone to interference and transmission errors. To reduce errors, both types of 802.11 automatically reduce the Physical layer data rate. IEEE 802.11b has three lower data rates (5.5, 2, and 1Mbit/sec), and 802.11a has seven (48, 36, 24, 18, 12, 9, and 6Mbits/sec). Higher (and more) data rates aren't 802.11a's only advantage. It also uses a higher frequency band, 5GHz, which is both wider and less crowded than the 2.4GHz band that 802.11b shares with cordless phones, microwave ovens, and Bluetooth devices.

The wider band means that more radio channels can coexist without interference. Each radio channel corresponds to a separate network, or a switched segment on the same network. One big disadvantage is that it is not directly compatible with 802.11b, and requires new bridging products that can support both types of networks. Other clear disadvantages are that 802.11a is only available in half the bandwidth in Japan (for a maximum of four channels), and it isn't approved for use in Europe, where HiperLAN2 is the standard.

IEEE 802.11g

Though 5GHz has many advantages, it also has problems. The most important of these is compatibility. The different frequencies mean that 802.11a products aren't interoperable with the 802.11b base. To get around this, the IEEE developed 802.11g, which should extend the speed and range of 802.11b so that it's fully compatible with the older systems.

The standard operates entirely in the 2.4GHz frequency, but uses a minimum of two modes (both mandatory) with two optional modes [Wireless Standards Up in the Air]. The mandatory modulation/access modes are the same CCK (Complementary Code Keying) mode used by 802.11b (hence the compatibility) and the OFDM (Orthogonal Frequency Division Multiplexing) mode used by 802.11a (but in this case in the 2.4GHz frequency band). The mandatory CCK mode supports 11Mbps and the OFDM mode has a maximum of 54Mbps. There are also two modes that use different methods to attain a 22Mbps data rate--PBCC-22 (Packet Binary Convolutional Coding, rated for 6 to 54Mbps) and CCK-OFDM mode (with a rated max of 33Mbps).

The obvious advantage of 802.11g is that it maintains compatibility with 802.11b (and 802.11b's worldwide acceptance) and also offers faster data rates comparable with 802.11a. The number of channels available, however, is not increased, since channels are a function of bandwidth, not radio signal modulation - and on that score, 802.11a wins with its eight channels, compared to the three channels available with either 802.11b or 802.11g. Another disadvantage of 802.11g is that it also works in the 2.4 GHz band and so due to interference it will never be as fast as 802.11a.

1.3. Mobile Networks: UMTS background

Universal Mobile Telecommunications System (UMTS) is envisioned as the successor to Global System for Mobile Communications (GSM). UMTS signals the move into the third generation (3G) of mobile networks. UMTS also addresses the growing demand of mobile and Internet applications for new capacity in the overcrowded mobile communications sky. The new network increases transmission speed to 2 Mbps per mobile user and establishes a global roaming standard.

UMTS, also referred to as Wideband Code Division Multiple Access (W–CDMA), is one of the most significant advances in the evolution of telecommunications into 3G networks. UMTS allows many more applications to be introduced to a worldwide base of users and provides a vital link between today's multiple GSM systems and the ultimate single worldwide standard for all mobile telecommunications, International Mobile Telecommunications–2000 (IMT–2000)

2G to 3G: GSM Evolution

Phase 1 of the standardization of GSM900 was completed by the European Telecommunications Standards Institute (ETSI) in 1990 and included all necessary definitions for the GSM network operations. Several tele-services and bearer services have been defined (including data transmission up to 9.6 kbps), but only some very basic supplementary services were offered. As a result, GSM standards were enhanced in Phase 2 (1995) to incorporate a large variety of supplementary services that were comparable to digital fixed network Integrated Services Digital Network (ISDN) standards. In 1996, ETSI decided to further enhance GSM in annual Phase 2+ releases that incorporate 3G capabilities.

GSM Phase 2+ releases have introduced important 3G features such as Intelligent Network (IN) services with Customized Application for Mobile Enhanced Logic (CAMEL), enhanced speech COmpression/DECompression (CODEC), Enhanced Full Rate (EFR), and Adaptive MultiRate (AMR), high–data rate services and new transmission principles with high-speed circuit-switched data (HSCSD), General Packet Radio Service (GPRS), and Enhanced Data Rates for GSM

Evolution (EDGE). UMTS is a 3G GSM successor standard that is downward-compatible with GSM, using the GSM Phase 2+ enhanced core network.

IMT-2000

The main characteristics of 3G systems, known collectively as IMT–2000, are a single family of compatible standards that have the following characteristics:

- used worldwide,
- used for all mobile applications,
- support both packet-switched (PS) and circuit-switched (CS) data transmission,
- offer high data rates up to 2 Mbps (depending on mobility/velocity),
- offer high spectrum efficiency.

IMT–2000 is a set of requirements defined by the International Telecommunications Union (ITU). As previously mentioned, IMT stands for International Mobile Telecommunications, and "2000" represents both the scheduled year for initial trial systems and the frequency range of 2000 MHz (WARC'92: 1885–2025 MHz and 2110–2200 MHz). All 3G standards have been developed by regional Standards Developing Organizations (SDOs). In total, proposals for 17 different IMT–2000 standards were submitted by regional SDOs to ITU in 1998—11 proposals for terrestrial systems and 6 for Mobile Satellite Systems (MSSs). Evaluation of the proposals was completed at the end of 1998, and negotiations to build a consensus among differing views were completed in mid 1999. All 17 proposals have been accepted by ITU as IMT–2000 standards.

The specification for the Radio Transmission Technology (RTT) was released at the end of 1999. The most important IMT–2000 proposals are the UMTS (W-CDMA) as the successor to GSM, CDMA2000 as the interim standard '95 (IS–95) successor, and time division–synchronous CDMA (TD–SCDMA) (universal wireless communication–136 [UWC–136]/EDGE) as TDMA (Time Division Multiple Access)–based enhancements to D–AMPS (Digital Advanced Mobile Phone System)/GSM—all of which are leading previous standards toward the ultimate goal of IMT–2000.

UMTS allows many more applications to be introduced to a worldwide base of users and provides a vital link between today's multiple GSM systems and IMT–2000. The new network also addresses the growing demand of mobile and Internet applications for new capacity in the overcrowded mobile communications sky. UMTS increases transmission speed to 2 Mbps per mobile user and establishes a global roaming standard.

UMTS is being developed by Third-Generation Partnership Project (3GPP), a joint venture of several SDOs—ETSI (Europe), Association of Radio Industries and Business/Telecommunication Technology Committee (ARIB/TTC) (Japan), American National Standards Institute (ANSI) T-1 (USA), telecommunications technology association (TTA) (South Korea), and Chinese Wireless Telecommunication Standard (CWTS) (China). To reach global acceptance, 3GPP is introducing UMTS in phases and annual releases. The first release (UMTS Rel. '99), introduced in December of 1999, defines enhancements and transitions for existing GSM networks. For the second phase (UMTS Rel. '00), similar transitions are being proposed as enhancements for IS–95 (with CDMA2000) and TDMA (with TD–CDMA and EDGE).

The most significant change in Rel. '99 is the new UMTS Terrestrial Radio Access (UTRA), a W–CDMA radio interface for land-based communications. UTRA supports time division duplex (TDD) and Frequency Division Duplex (FDD). The TDD mode is optimized for public micro and pico cells and unlicensed cordless applications. The FDD mode is optimized for wide-area coverage, i.e., public macro and micro cells. Both modes offer flexible and dynamic data rates up to 2 Mbps. Another newly defined UTRA mode, multicarrier (MC), is expected to establish compatibility between UMTS and CDMA2000.

UMTS Network architecture

UMTS (Rel. '99) incorporates enhanced GSM Phase 2+ Core Networks with GPRS and CAMEL. This enables network operators to enjoy the improved cost/efficiency of UMTS while protecting their 2G investments and reducing the risks of implementation.

In UMTS release 1 (Rel. '99), a new radio access network UMTS terrestrial radio access network (UTRAN) is introduced. UTRAN, the UMTS Radio Access Network (RAN), is connected via the Iu to the GSM Phase 2+ core network (CN). The Iu is the UTRAN interface between the Radio Network Controller (RNC) and CN; the UTRAN interface between RNC and the Packet-Switched domain of the CN (Iu–PS) is used for PS data and the UTRAN interface between RNC and the Circuit Switched domain of the CN (Iu–CS) is used for CS data. "GSM–only" Mobile Stations (MSs) will be connected to the network via the GSM air (radio) interface (Um). UMTS/GSM dual-mode User Equipment (UE) will be connected to the network via UMTS air (radio) interface (Uu) at very high data rates (up to almost 2 Mbps). Outside the UMTS service area, UMTS/GSM UE will be connected to the network at reduced data rates via the Um.

Maximum data rates are 115 kbps for CS data by HSCSD, 171 kbps for PS data by GPRS, and 553 kbps by EDGE. Handover between UMTS and GSM is supported, and handover between

UMTS and other 3G systems (e.g., multicarrier CDMA [MC–CDMA]) will be supported to achieve true worldwide access.

The Public Land Mobile Network (PLMN) described in UMTS Rel. '99 incorporates three major categories of network elements:

- GSM Phase 1/2 core network elements: Mobile Services Switching Center (MSC), Visitor Location Register (VLR), Home Location Register (HLR), Authentication Center (AC), and Equipment Identity Register (EIR);
- GSM Phase 2+ enhancements: GPRS (Serving GPRS Support Node [SGSN] and Gateway GPRS Support Node [GGSN]) and CAMEL (CAMEL Service Environment [CSE]);
- UMTS specific modifications and enhancements, particularly UTRAN.

The GSM Phase 1/2 PLMN consists of three subsystems: the Base Station Subsystem (BSS), the Network and Switching Subsystem (NSS), and the Operations Support System (OSS). The BSS consists of the functional units: Base Station Controller (BSC), Base Transceiver Station (BTS) and Transcoder and Rate Adapter Unit (TRAU). The NSS consists of the functional units: MSC, VLR, HLR, EIR, and the AC. The MSC provides functions such as switching, signalling, paging, and inter–MSC handover. The OSS consists of Operation and Maintenance Centers (OMCs), which are used for remote and centralized Operation, Administration, and Maintenance (OAM) tasks.

The most important evolutionary step of GSM toward UMTS is GPRS. GPRS introduces PS into the GSM CN and allows direct access to Packet Data Networks (PDNs). This enables high–data rate PS transmission well beyond the 64 kbps limit of ISDN through the GSM CN, a necessity for UMTS data transmission rates of up to 2 Mbps. GPRS prepares and optimizes the CN for high–data rate PS transmission, as does UMTS with UTRAN over the RAN. Thus, GPRS is a prerequisite for the UMTS introduction. Two functional units extend the GSM NSS architecture for GPRS PS services: the GGSN and the SGSN. The GGSN has functions comparable to a Gateway MSC (GMSC). The SGSN resides at the same hierarchical level as a Visited MSC (VMSC)/VLR and therefore performs comparable functions such as routing and mobility management.

CAMEL enables worldwide access to operator-specific IN applications such as prepaid, call screening, and supervision. CAMEL is the primary GSM Phase 2+ enhancement for the introduction of the UMTS Virtual Home Environment (VHE) concept. VHE is a platform for flexible service definition (collection of service creation tools) that enables the operator to modify or enhance existing services and/or define new services. Furthermore, VHE enables worldwide access to these operator-specific services in every GSM and UMTS PLMN and introduces location-based services (by interaction with GSM/UMTS mobility management). A CSE and a new Common Control Signalling System 7 (SS7) (CCS7) protocol, the CAMEL Application Part (CAP), are required on the CN to introduce CAMEL.

As mentioned above, UMTS differs from GSM Phase 2+ mostly in the new principles for air interface transmission (W–CDMA instead of time division multiple access [TDMA]/frequency division multiple access [FDMA]). Therefore, a new RAN called UTRAN must be introduced with UMTS. Only minor modifications, such as allocation of the TransCoder (TC) function for speech compression to the CN, are needed in the CN to accommodate the change. The TC function is used together with an InterWorking Function (IWF) for protocol conversion between the A and the Iu–CS interfaces.

The description of the general UTMS architecture can be found in [3G2310] and deeply in [3G238], while the description of the network elements are reported in [3G238] and [3G230].

UTRAN

The UMTS standard can be seen as an extension of existing networks. Two new network elements are introduced in UTRAN, RNC, and Node B. UTRAN is subdivided into individual Radio Network Systems (RNSs), where each RNS is controlled by an RNC. The RNC is connected to a set of Node B elements, each of which can serve one or several cells.

Existing network elements, such as MSC, SGSN, and HLR, can be extended to adopt the UMTS requirements, but RNC, Node B, and the handsets must be completely new designs. RNC will become the replacement for BSC, and Node B fulfils nearly the same functionality as BTS. GSM and GPRS networks will be extended, and new services will be integrated into an overall network that contains both existing interfaces such as A, Gb, and Abis, and new interfaces that include Iu, UTRAN interface between Node B and RNC (Iub), and UTRAN interface between two RNCs (Iur).

UMTS defines four new open interfaces:

- Uu: UE to Node B (UTRA, the UMTS W–CDMA air interface);
- Iu: RNC to GSM Phase 2+ CN interface (MSC/VLR or SGSN):
    - Iu-CS for circuit-switched data,
    - Iu-PS for packet-switched data;
- Iub: RNC to Node B interface;
- Iur: RNC to RNC interface, not comparable to any interface in GSM.

The Iu, Iub, and Iur interfaces are based on ATM transmission principles.

The RNC enables autonomous Radio Resource Management (RRM) by UTRAN. It performs the same functions as the GSM BSC, providing central control for the RNS elements (RNC and Node Bs).

The RNC handles protocol exchanges between Iu, Iur, and Iub interfaces and is responsible for centralized Operation and Maintenance (O&M) of the entire RNS with access to the OSS. Because the interfaces are ATM–based, the RNC switches ATM cells between them. The user's circuit-switched and packet-switched data coming from Iu–CS and Iu–PS interfaces are multiplexed together for multimedia transmission via Iur, Iub, and Uu interfaces to and from the UE.

The RNC uses the Iur interface, which has no equivalent in GSM BSS, to autonomously handle 100 percent of the RRM, eliminating that burden from the CN. Serving control functions such as admission, RRC connection to the UE, congestion and handover/macro diversity are managed entirely by a single Serving RNC (SRNC).

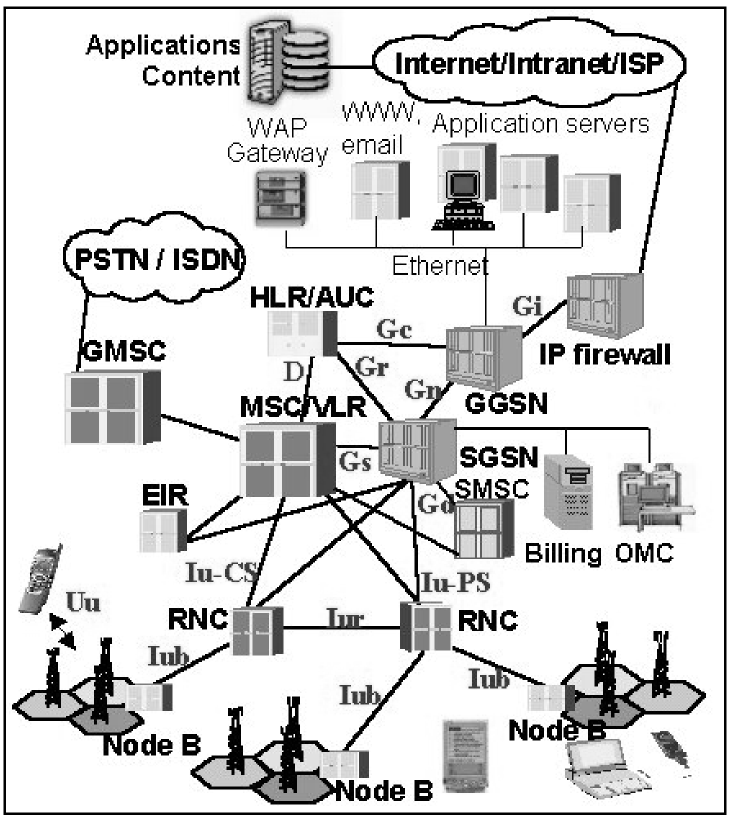

Fig. I.8. UMTS network layout example

If another RNC is involved in the active connection through an inter–RNC soft handover, it is declared a Drift RNC (DRNC). The DRNC is only responsible for the allocation of code resources. A reallocation of the SRNC functionality to the former DRNC is possible (Serving Radio Network Subsystem [SRNS] relocation).

The term Controlling RNC (CRNC) is used to define the RNC that controls the logical resources of its UTRAN access points.

The Fig. I.8 shows how an UMTS 3G network could be build.

Node B

Node B is the physical unit for radio transmission/reception with cells.

Depending on sectoring (omni/sector cells), one or more cells may be served by a Node B. A single Node B can support both FDD and TDD modes, and it can be co-located with a GSM BTS to reduce implementation costs. Node B connects with the UE via the W–CDMA Uu radio interface and with the RNC via the Iub asynchronous transfer mode (ATM)–based interface. Node B is the ATM termination point.

The main task of Node B is the conversion of data to and from the Uu radio interface, including Forward Error Correction (FEC), rate adaptation, W–CDMA spreading/despreading, and Quadrature Phase Shift Keying (QPSK) modulation on the air interface. It measures quality and strength of the connection and determines the Frame Error Rate (FER), transmitting these data to the RNC as a measurement report for handover and macro diversity combining. The Node B is also responsible for the FDD softer handover. This micro diversity combining is carried out independently, eliminating the need for additional transmission capacity in the Iub.

The Node B also participates in power control, as it enables the UE to adjust its power using DownLink (DL) Transmission Power Control (TPC) commands via the inner-loop power control on the basis of uplink (UL) TPC information. The predefined values for inner-loop power control are derived from the RNC via outer-loop power control.

1.4. Genetic Algorithms background

According to [H94] "Evolutionary algorithm is an umbrella term used to describe computer-based problem solving systems which use computational models of evolutionary processes as key elements in their design and implementation". A variety of evolutionary algorithms have been proposed. The major ones are:

- genetic algorithms,
- evolutionary programming,
- evolution strategies,
- classifier system,

- genetic programming.

They all share a common conceptual base of simulating the evolution of individual structures via processes of selection, mutation and reproduction. The processes depend on the perceived performance of the individual structures as defined by an environment. More precisely, evolutionary algorithms maintain a population of structures, that evolve according to rules of selection and other operators, that are referred to as “search operators”, (or genetic operators), such as recombination and mutation. Each individual in the population receives a measure of its fitness in the environment. Reproduction focuses attention on high fitness individuals, thus exploiting the available fitness information. Recombination and mutation perturb those individuals, providing general heuristics for exploration. Although simplistic from a biologist’s viewpoint, these algorithms are sufficiently complex to provide robust (good performance across a variety of problem types) and powerful adaptive search mechanisms.

Introduction to GA

Genetic algorithms originated from the studies of cellular automata, conducted by John Holland and his colleagues at the University of Michigan. Holland’s book [H92], published in 1975, is generally acknowledged as the beginning of the research of genetic algorithms. Until the early 1980s, the research in genetic algorithms was mainly theoretical, with few real applications. This period is marked by ample work with fixed length binary representation in the domain of function optimisation by, among others, De Jong and Hollstien. Hollstien’s work provides a careful and detailed analysis of the effect that different selection and mating strategies have on the performance of a genetic algorithm. De Jong’s work attempted to capture the features of the adaptive mechanisms in the family of genetic algorithms that constitute a robust search procedure.

From the early 1980s the community of genetic algorithms has experienced an abundance of applications which spread across a large range of disciplines. Each and every additional application gave a new perspective to the theory. Furthermore, in the process of improving performance as much as possible via tuning and specialising the genetic algorithm operators, new and important findings regarding the generality, robustness and applicability of genetic algorithms became available.

Following the last couple of years of furious development of genetic algorithms in the sciences, engineering and the business world, these algorithms in various guises have now been successfully applied to optimisation problems, scheduling, data fitting and clustering, trend spotting and path finding.

As [H94], [D95], and [G89] stated, the genetic algorithm is a model of machine learning which derives its behaviour from a metaphor of the processes of evolution in nature. This is done by the creation within a machine of a population of individuals represented by chromosomes, in essence a set of character strings that are analogous to the base-4 chromosomes that can be seen in our own DNA. The individuals in the population then go through a process of evolution which is, according to Darwin, made up of the principles of mutation and selection; however, the modern biological evolution theory also knows crossover and isolation mechanisms to improve the adaptiveness of the living beings to their environments.

With genetic algorithms, elements or chunks of elements are swapped between individuals as if by sexual combination and reproduction (crossover), others are changed at random (mutation). New generations appear from clones of the current population, in proportion to their fitness: a single objective function of the parameters that returns a numerical value, to distinguish between good and bad solutions. Fitness is then used to apply selection pressure to the population in a 'Darwin' fashion (survival of the fittest).

GAs differ from more normal optimisation and search procedures in four ways:

- GAs work with a coding of the parameter set, not the parameters themselves;
- GAs search from a population of points, not a single point;
- GAs use payoff (objective function) information, not derivatives or other auxiliary knowledge;
- GAs use probabilistic transition rules, not deterministic rules.

Genetic algorithms require the natural parameter set of the optimisation problem to be coded as a finite-length string (analogous to chromosomes in biological systems) containing characters, features or detectors (analogous to genes), taken from some finite-length alphabet. Usually, the binary alphabet that consists of only 0 and 1 is taken. Each feature takes on different values (alleles) and may be located at different positions (loci). The total package of strings is called a structure or population (or, genotype in biological systems). A summary of the correspondence between natural and artificial terminology is given in Table I.1.

Table I.1. Comparison of Natural and GA Terminology

Chromosome	String
Gene	feature, character or detector
Allele	feature value
Locus	string position
Genotype	structure, or population
Phenotype	parameter set, alternative solution, a decoded structure

Structure of a GA

A pseudo code gives an abstract view of an algorithm. With genetic algorithms, this pseudo code is very simple; see Fig. I.9 and Fig. I.10.

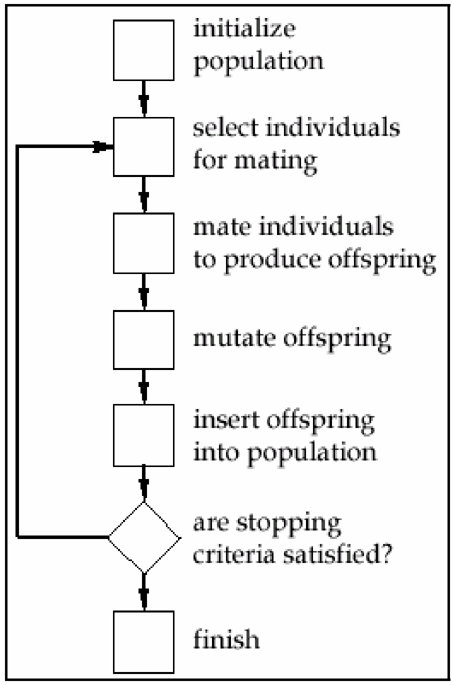

Fig. I.9. Standard genetic algorithm process

***Procedure**: Genetic Algorithms*
Begin
    g := 0;
    initialize P(g);   *{P(g) is the population of P_{Dim} individuals in generation g}*
    While** (not termination condition) **do
        Begin
            evaluate P(g);
            select best P(g);   *{$P_{best}(g)$ is the most fitting part of P(g) }*
            discard worst P(g);   *{$P_{worst}(g) = P(g) - P_{best}(g)$}*
            cross-over $P_{best}(g)$;   *{$P_{cr}(g)$ is the cross-over of $P_{best}(g)$}*
            mutation $P_{cr}(g)$;   *{$P_{mut}(g)$ is the mutation of $P_{best}(g)$}*
            $P(g)=P_{best}(g) + P_{mut}(g)$;   *{$P_{dim}$ does not change because $|P_{mut}(g)| = |P_{worst}(g)|$}*
            g = g+1;
        End
End

Fig. I.10. General structure of a genetic algorithm

A genetic algorithm starts with a population of strings to be able to generate successive populations of strings afterwards. The initialisation is usually done randomly. This means, with binary strings for example, that every allele is set to 0 or 1, with each value having a chance of 50% to occur. This can be achieved by simulating the tossing of a coin with head=1 and tail=0.

From now on, a simple example will be used to demonstrate every feature of the genetic algorithm. This example will be: finding the maximum of the function $u(x,y) = (x-7)^2 + (y-3)^2$ with both x and y on an integer interval [0,7], see Fig. I.11.

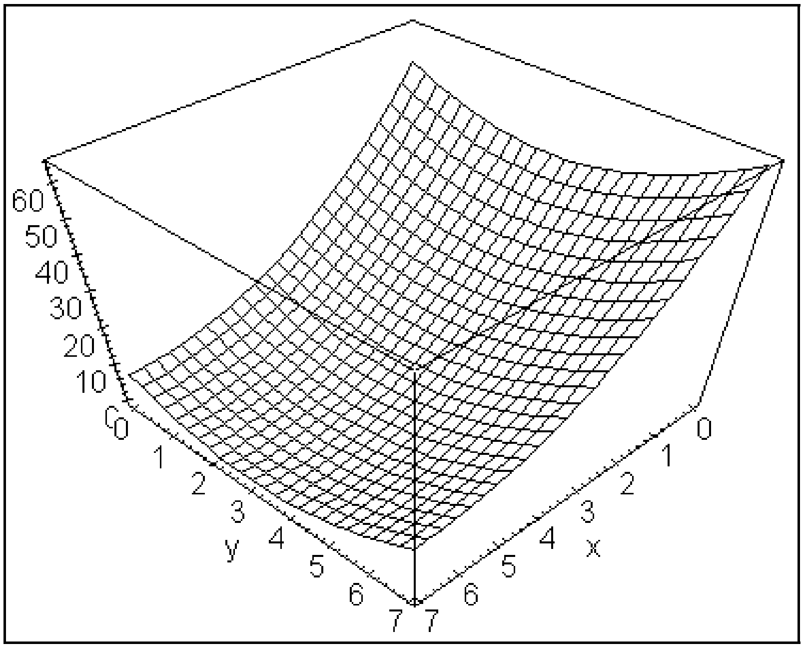

Fig. I.11. Example for GA.

The first thing to do, is generate a code for this problem. A binary representation of six bits has been chosen, where the three bits on the left represent the x value and the three bits on the right the y value. For example 011101 means: x=011b=3 and y=101b=5. Table I.2 shows a possible randomly generated population of four six-bit individuals using successive coin flips.

Table I.2. Example of an Initial Population

no.	String	
1.	100001	(x=4, y=1)
2.	001100	(x=1, y=4)
3.	110010	(x=6, y=2)
4.	000100	(x=0, y=4)

After every generated population, every individual of the population must be evaluated to be able to distinguish between good and bad individuals. This is done by mapping the objective function to a ‘fitness function’: a non-negative figure of merit. This can be done in the following ways [G89]:

- when the objective is maximisation of a utility or profit function u(x) the problem of negative u(x) values is overcome by transforming fitness according to the equation:

  $f(x) = u(x) + C_{min}$ when $u(x) + C_{min} > 0$,

$f(x) = 0$ otherwise.

- when the objective is minimisation of a cost function $g(x)$ the minimisation problem has to be transformed to a maximisation problem and assure that the measure is non-negative by using the following cost-to-fitness transformation:

  $f(x) = C_{max} - g(x)$ when $C_{max} - g(x) > 0$,

  $f(x) = 0$ otherwise.

C_{min} or C_{max} may be chosen as an input coefficient, as the absolute value of the worst u-value respectively the largest g-value in the current or last k generations, or as a function of the population variance.

With this example function, of which is has to be found the maximum, it can be chosen $C_{min}= 0$ because the objective function $u(x)$ will never be negative. Thus $f(x) = u(x)$ (fitness is evaluation). Let's take a look at the evaluation of the four initial strings (Table I.3):

Table I.3. Evaluation of the Initial Population

no.	String	(x,y)	fitness
1.	100001	(4,1)	13
2.	001100	(1,4)	37
3.	110010	(6,2)	2
4.	000100	(0,4)	50

An important aspect is to decide which individuals should be chosen as parents for the purpose of procreation. With genetic algorithms, this selection is based on the string fitness: according to the 'survival of the fittest' principle, a 'string A' that is twice as fit as a 'string B' would be expected to appear twice as much in the next generation. Note that the genetic algorithm does not select strings directly by their rank in the population, so the best string is not guaranteed to be a parent.

The way to implement this kind of reproduction into a simple genetic algorithm is to create a biased roulette wheel where each current string in the population has a roulette wheel slot sized in proportion to its fitness. To reproduce, let's simply spin the weighted roulette wheel as many times as the population size (four times in this example) [G89]. By dividing an individual's fitness value with the average of all fitness values, it is possible to calculate the expected count of this individual in the next generation. The average of all the fitness functions of this example is 25.5, so the expected count of individual one in the next generation is 13 / 25.5 = 0.51. Other expected counts are shown in table 3.3, as well as the normalised fitness values, which are equal to the fitness values divided by the total sum of all fitness values (102 in this example), multiplied by 100%. The normalised fitness gives the chance of an individual to be chosen as a parent. A method to actually select an individual as a parent is to use a sum function $S_i = \sum_{j=1}^{i} f_i$, which gives the sum of all fitness values from individual one to individual *i,* and randomly and uniformly choose an integer between 0 and the sum of all fitness values. The first individual whose S_i is equal or greater than this integer will be chosen as a parent. The S_i values are shown in Table I.4. For example, suppose the randomly chosen number is 53 then individual 4 will be chosen as a parent because S_4 is the first value that succeeds 53. This routine will be repeated until there are 4 parents. The parents are also shown in Table I.4

Table I.4. Reproduction Results

No.	string	(x,y)	fitness	Normalised	S_i	expected count	actual
1.	100001	(4,1)	13	12.7%	13	0.51	1
2.	001100	(1,4)	37	36.3%	50	1.45	1
3.	110010	(6,2)	2	02.0%	52	0.08	0
4.	000100	(0,4)	50	49.0%	102	1.96	2

Once two parents have been selected, the genetic algorithm combines them to create two new offspring. Combination is performed by the crossover operator. Many different crossover operators exist as two-point, uniform, partially mixed, and uniform order-based crossover. In this background, the simple, or one-point crossover has been described only. This operator randomly and uniformly selects an integer k between 1 and the string length less one [1, *l*-1]. Two new strings are created by swapping all characters between positions k and l, see Fig. I.12.

Before crossover	After crossover
A B C D|E F G H	A B C D|M N O P
I J K L |M N O P	I J K L |E F G H

Fig. I.12. The One-Point Crossover Operator

The role of the crossover operator is to allow the advantageous traits to be spread throughout the population in order that the population as a whole may benefit from this chance discovery. When crossover is left out the system uses mutation to search the area in the neighbourhood of a number of points without the capability of combining the best features of those points, which leads to diminished performance [D91]. Note that the crossover operator is not always performed : a biased coin is flipped with a probability *CROSS* (often taken as an input coefficient) to decide whether or not crossover occurs. The crossover is the prime distinguishing factor of a genetic algorithm from other optimisation algorithms.

The last operator in the simple genetic algorithm is the mutation algorithm. How important its role is, is still a matter of debate. Most people however, believe that mutation plays a secondary role in the simple genetic algorithm. The effect of mutation is to reintroduce divergence into a converging population. In the latter stages of a genetic algorithm run the algorithm may be converging upon a local maximum. By mutating some chromosomes it is possible to find a way past this local maximum. The biological inspiration behind this operator is the way in which a chance mutation in a natural chromosome can lead to the development of desirable traits which give the individual displaying these characteristics an advantage over its competitors.

The mutating operator simply tosses a biased coin with probability *MUT* (which is very small) at each bit and, according to that result, changes a 1 into a 0 and vice versa, see Fig. I.13.

Before mutation	After mutation
1 0 **1** 1 0 0	1 0 **0** 1 0 0

Fig. I.13. The Mutation Operator

Let's suppose that in this example, only bit 4 (the second bit from the left) is being changed through mutation, see Fig. I.13. The new population is shown in Table I.5, as well as the new fitness values.

Table I.5. New Population and Fitness after Crossover and Mutation

no.	selected parents	after crossover	after mutation	new fitness
1.	10|0001	101100	10**0**100	10
2.	00|1100	000001	000001	53
3.	00010|0	000100	000100	50
4.	00010|0	000100	000100	50

Then, a new string with high fitness has appeared. The sum of the fitness values has increased from 102 to 163 and the average has increased from 25.5 to 40.8, and all this in one generation. Following the reproduction, crossover and mutation, it can be seen that of the initial population, strings 1 and 2 were selected ones (average fitness), string 3 was not selected (low fitness) and string 4 was selected twice (high fitness). Crossover provided the high-fitness string 000001 (string 2) but also the low-fitness string 101100 (string 1) in which a mutation took place which, in this case, increased the fitness.

The simple genetic algorithm is a powerful tool that is able to converge rapidly to an optimum of many different objective functions (in the example a two-variable function, but a function of more variables is easy to implement). The user has to create a code scheme, a fitness function and implement these into the genetic algorithm, whose mechanisms are easy to implement into a computer program.

Chapter II

Literature Survey

This Chapter is intended for introducing the reader into the literature state-of-art about the problems and research fields that will be faced in the following Chapters.

The following sections provide brief descriptions on the use of GA algorithm for solving network design problem in multicast environment, the power management standard in wireless networks, and the QoS management in 3G networks.

2.1. Network design for multicast grouping

In the last years, new applications with multipoint data transmissions, such as resource discovery, multimedia conferencing, distant learning, and video on demand, came up. These applications are expected to become of key importance in the next future for the network service providers, which will need to adapt their existing networks so as to appropriately accommodate the additional multicast traffic. In this framework, a link capacity assignment problem has to be addressed based on the deployed network topology, the existing unicast traffic, and the predicted multicast traffic. The huge number of multiple routes between each source-destination pair to be analyzed makes this problem quite complex, requiring an appropriate heuristic solution to restrict the computational complexity. In this section, it is presented a solution that is based on the promising genetic algorithms (GA) and relies on an approach similar to that proposed in [STV01] that was aimed at the design of communication networks in case of unicast traffic.

In the past, the network dimensioning problem for multicast traffic, but without the shortest path routing constraints, has previously been studied by Prytz and Forsgren [PF01]. A similar problem, also without the shortest path routing constraints, has independently been studied by Bienstock and Bley [BB00]. Related problems on multicast network design have been studied by Leung and Yung [LY98] and Chen *et al.* [CGY00]. Leung and Yung [LY98] consider the multicast routing problem for a single multicast group, but with an extra total delay requirement between the routers. Chen *et al.* [CGY00] consider packing several multicast groups in an existing network such that the maximum congestion on any link is minimized.

The network design problem with shortest path routing constraints has recently been studied by Holmberg and Yuan [HY01]. They treat unicast, point-to-point traffic and explicitly include the possibility to use equal-cost multipaths in the model. Several heuristic methods are proposed to find good feasible solutions. The traffic engineering problem of finding shortest path routing weights in an existing network such that link congestion is minimized, has been studied by Fortz and Thorup [FT00]. They show that it is possible to achieve nearly optimal load balancing in a network by selecting link routing weights different from the ones given by the inverse-capacity heuristic. They also present a local search heuristic for finding a good set of link routing weights.

Recently, Ericsson, Resende and Pardalos [ERP01] have described a genetic algorithm for the same problem. Staehle, Küohler and Kohlhaas [SKK00] have considered optimizing link routing weights to minimize the maximum utilization on any link. Lin and Wang [LW93] have studied a similar problem which also included multicast traffic. They suggest some primal heuristics together with a Lagrangian relaxation lower bounding method. In [PF01], a new approach for the dimensioning of the backbone capacity for new networks to accommodate multiple multicast routing trees has been proposed; while, in [CGY00], the authors consider packing existing multicast groups in a network such that the maximum congestion on any link is minimized.

Network capacity assignment problem

The addressed problem can be summarized as follows: given an existing network with known unicast traffic, find the optimal link capacity assignment to accommodate the multicast traffic generated by a certain group of new multicast sources. Accordingly, the network topology and the underlying unicast traffic are known as well as the expected multicast traffic. The existing network can be modelled by a directed graph $G(\mathbf{V},\mathbf{E})$, where **V** (|**V**|=*N*) is the set of communication nodes n_i and **E** (|**E**|=*H*) is the set of communication links l_{ij}. l_{ij} represents a directed edge from n_i to n_j with an available bandwidth b_{ij} (link capacity minus the existing unicast traffic) that may be different from b_{ji}. Let **S** be a subset of **V** denoting the group of multicast source nodes: $\mathbf{S}\subset\mathbf{V}$, |**S**|=*M*, and *M*<*N*. A source node s_z transmits multicast data to every node belonging to its destination group $\mathbf{D}_z$ at rate y_z: $\mathbf{D}_z \subset \mathbf{V}$, $|\mathbf{D}_z| = K_z$, and $z = 1,\dots,M$.

The objective of this capacity assignment problem is equivalent to find the optimal set of traffic distribution trees on the basis of an appropriate cost function. Let's represent a possible solution *h* by the set of multicast trees $\mathbf{F}_h = \{\mathbf{T}_{h1},\dots,\mathbf{T}_{hM}\}$, where each tree $\mathbf{T}_{hz}$ is rooted at source node s_z and has to reach all the nodes in $\mathbf{D}_z$. The number of nodes involved in $\mathbf{F}_h$ is N_h. On the basis of the above notation, the total cost associated to a solution *h* is defined as follows:

$$C_h = \sum_{i=1}^{N}\sum_{j=1}^{N}\left[A(y_{hij}, b_{ij}) \cdot c_1 + U(y_{hij}, b_{ij}) \cdot c_2\right] + N_h \cdot c_3 + \sum_{k=1}^{N_h} y_{hk} \cdot c_4 \, , \tag{2.1}$$

where:

$$y_{hij} = \sum_{z=1}^{M} y_z \cdot x_{zij} \qquad \text{with} \begin{cases} x_{zij} = 1 & \text{if } l_{ij} \in \mathbf{T}_{hz} \\ x_{zij} = 0 & \text{otherwise} \end{cases}, \tag{2.2}$$

$$y_{hk} = \sum_{z=1}^{M} y_z \cdot x_{zk} \qquad \text{with} \begin{cases} x_{zk} = 1 & \text{if } n_k \in \mathbf{T}_{hz} \\ x_{zk} = 0 & \text{otherwise} \end{cases}, \tag{2.3}$$

$$A(a,b) = \begin{cases} a & \text{if } a < b \\ b & \text{otherwise} \end{cases}, \tag{2.4}$$

$$U(a,b) = \begin{cases} a-b & \text{if } a \geq b \\ 0 & \text{otherwise} \end{cases}. \tag{2.5}$$

y_{hij} represents the total amount of bandwidth required by solution h in link l_{ij}, that is divided in the portion satisfied by the available capacity b_{ij} (computed by means of function $A(a,b)$) and the remaining part for which link upgrades are required (computed by means of function $U(a,b)$). The c_1 and c_2 coefficients in the first addend of (2.1) are used to differently weight the bandwidth requirements addressed by available capacity and link upgrades, respectively. Usually, c_2 should be taken greater than c_1. Differently, the second addend in (2.1) is related to the costs of O&M (Operation and Maintenance) of multicast functionalities for each node involved in solution $\mathbf{F}_h$: it is a fixed cost. The last addend in (2.1) is concerned with the amount of multicast data that have to be routed over the whole network in solution $\mathbf{F}_h$ (this is a variable cost). The operator can configure these cost coefficients c_i based on some considerations:

- ✓ A common need is to try accommodating the multicast sessions without introducing link capacity upgrades that usually are quite expensive. In this case, c_2 should be set quite higher than cost c_1, for example n times higher. It means that the operator prefers introducing upgrades only if alternative solutions without upgrades require distribution trees with some segments n times longer or more. The setting of these parameters should also take into account the quality of service to be provided: longer trees mean higher delays that have to be avoided in case of streaming applications. Additionally, with $c_2 = \infty$, the operator may prevent from considering solutions with upgrades. Anyway, not all the configurations (network topology, link capacity, multicast source rate, and unicast traffic) may have a solution without capacity upgrades; then, a very high value, but not infinite, for c_2 is usually recommended in these cases.

- ✓ The first addend in (2.1) is the most important since it is related to the link usage. To this, the resulting value should be always higher than the other addends in (2.1). This condition is assured by taking c_3 and c_4 not greater than c_1 and c_2. Indeed, c_3 and c_4 allow discriminating one solution respect to the others when different solutions have the same link usage cost. Let's consider two solutions *i* and *j* with equal link usage costs, different total multicast traffic $\left(\sum_{k=1}^{N_i} y_{ik} > \sum_{k=1}^{N_j} y_{jk}\right)$, and different number of involved nodes $(N_i < N_j)$. Solution *i* is selected if its total multicast traffic exceeds that of solution *j* no more than c_4/c_3 times the number of saved nodes respect to *j*. Accordingly, the ratio c_4/c_3 provides the maximum amount of additional multicast traffic that can be accepted in a solution for every saved multicast node.

Related works address the same problem with different constraints and/or configurations. In particular, the algorithm proposed by Chen *et al* [CGY00] focuses on accommodating multiple multicast trees in the same network without considering the possibility to upgrade link capacity. According to this algorithm, "congested" multicast trees are modified looking for alternative paths for the congested links. Indeed, this is the most useful practical configuration. A quite similar problem is solved by the algorithm proposed setting $c_2 = \infty$. In this case, the resulting solution may consist on large trees that impact on the transmission delay. To limit the delay, Chen *et al.* [CGY00]. bound the cost of each multicast tree to guarantee service quality. This feature is not available in the algorithm; at most, the c_3 and c_4 costs can be increased in order to try to reduce the tree extension. To compare the two configurations (possibility to introduce upgrade and not), in Fig. II.1 the solution in case of $c_2 = \infty$ is shown. The resulting solution consists on larger trees, as expected.

Proposed Solution

As explained in Chapter I, Section 1.4, the application of the GA algorithm requires the representation of the problem solution with a chromosome. A population of chromosome is created and the genetic operators, such as mutation and crossover, are applied to produce new offspring (see also Fig. I.10). The proposed application of the GA algorithm to the multicast capacity assignment problem relies on two phases: the search of a set of tree solutions for each multicast session alone; and the search of the best trees combination for all the multicast sessions together starting from the output of the previous phase.

As to the first phase, the search of the best trees for a single session *z* (source s_z and destination group $\mathbf{D}_z$) is based on the following steps:

1. Create an array $\mathbf{L}_i$ of unicast source-destination paths $s_z - d_i$ for every $d_i \in \mathbf{D}_z$, selected by using a depth-first search algorithm. The number of arrays $\mathbf{L}_i$ is equal to K_z $(i = 1,..., K_z)$ and the dimension of each array ($|\mathbf{L}_i| = \alpha_i$) is defined by limiting the maximum number of hops per path. Every resulting path for a destination d_i is referred to with its position *j* within the array $\mathbf{L}_i$.
2. Also referring to Fig. I.10, create a population $P(g = 0)$ of P_{Dim} strings (chromosomes), each composed by K_z characters (genes). Every character *i* represents the index of the path selected from $\mathbf{L}_i$, that can be a value in the range $(1,..., \alpha_i)$. From each chromosome a tree is extracted by grouping one-by-one all the paths associated to each gene and deleting all the duplicate sub-path routed at the source node.
3. Initialize the population randomly.
4. Perform the evolution of the population by applying crossover and mutation operators. In particular, the most fitting part of the population $P_{best}(g)$ is selected and directly inserted in the new generation, while the rest of the population $P_{worst}(g)$ is discarded and replaced by a sub-population $P_{cr}(g)$, that is composed by new individuals created by applying the crossover operator. Then, the mutation operator is applied to the chromosomes of the new population. A single point crossover operator is used to interchange the elements of two strings, while mutation operator tries to lead the search out of local optima. In the case of two identical chromosomes resulted after the crossover and mutation operations, two individuals are randomly generated. The evaluation of the chromosome fitness is performed by means of the following formula:

   $$f = 1 - \frac{C_h}{N \cdot c_3 + y_z \cdot (H \cdot c_2 + N \cdot c_4)} \tag{2.6}$$

   where C_h is the cost of the tree associated to the chromosome and computed by means of (2.1) with $M = 1$.
5. Repeat step '4' (GA evolves while increasing *g*) until the number of iterations *IT* reaches a threshold value.

During the first phase, the previous steps are performed for each multicast session. Then, at the end P_{Dim} trees for each session are obtained. The second phase relies on the same steps with two basic variants: the population is made by M characters, each one representing a multicast tree obtained from the previous phase and assuming a value in the range $(1,...,P_{Dim})$; the used fitness function is substituted with the following formula:

$$f = 1 - \frac{C_h}{N \cdot c_3 + \left(\sum_{z=1}^{M} y_z \right) \cdot (H \cdot c_2 + N \cdot c_4)} . \qquad (2.7)$$

When the GA algorithm has completed the selected number of iterations, the final multicast capacity assignment configuration for multiple multicast sessions is obtained.

Experiments

This subsection presents the results relevant to the application of the proposed method to a network $G(\mathbf{V},\mathbf{E})$ with N=20 and H=44. The background unicast traffic occupation for each link was randomly set in the range 40%-90% and the cost factors were set to the following values: $c_1 = c_3 = 1$, $c_2 = 1.5$, and $c_4 = 0.5$ (the bandwidth in (2.1) was expressed in Mbps). As to the GA settings, the following parameter values have been experimented: number of sources in the range (1,…,10), crossover percentage from 30% to 70%, mutation probability varying from 0.01 to 0.1, number of destination nodes varying from 5 to 7, population size in the range (8,16,32,64), and a number of iterations equals to 1000.

Since the problem of accommodating multiple multicast trees is NP-complete and the minimal cost solution cannot be determined without taking all the links into consideration, only approximations of the optimal solution can be found. Then, to evaluate the performance of the proposed GA-based algorithm, the results are compared with a heuristic "quasi-exhaustive" search algorithm, named LB (lower bound). It does not guarantee to obtain the solution with the minimum cost (optimal solution) and its accuracy depends on the number of trees taken in $\mathbf{G}_z$ ($|\mathbf{G}_z|$). However, the LB solution is considered quite close to the optimal since a high value for $|\mathbf{G}_z|$ is used. In particular, this is obtained by evaluating the LB solution at increasing values of $|\mathbf{G}_z|$ (up to 100) and selecting $|\mathbf{G}_z|$ over which the obtained solutions remain unchanged. This does not assure that the optimal solution is reached but it is a good indication. That is the reason why it was called lower bound algorithm.

For the test network (20 nodes and 44 links), $|\mathbf{G}_z|$=18 is used; by changing the network and the unicast traffic different values were found. It was observed, as expected, greater values for $|\mathbf{G}_z|$ when increasing the number of nodes in the network.

The search of a lower bound is performed as follows: a multicast session *z* is considered together with the associated set $\mathbf{U}_z = s_z \cup \mathbf{D}_z$ of nodes; all Steiner trees with a given maximum number of hops are generated for this set of nodes by using the heuristic algorithm called TM heuristic [TM80], without taking the bandwidth constraints into consideration; a set $\mathbf{G}_z$ of the best multicast trees created in the previews step is selected by evaluating the cost of every Steiner tree with equation (2.1) and *M*=1; the previous steps are performed for *z*=1,...,*M*; all the possible combinations of *M* Steiner trees (one for each set $\mathbf{G}_z$) are considered and the one that minimizes C_h in (2.1) is considered as the lower bound.

The proposed algorithm has found optimal results by considering a population size equal to 64 chromosomes, crossover percentage equal to 50%, and a mutation probability equal to 0.01. Table II.1 presents the obtained results together with those relevant to the lower bound search in terms of cost reduction (reduction of the cost function respect to the initial population) and the processing time (Pentium III, 1.8GHz, 512MB RAM memory).

As to the LB and GA-based algorithms, the complexity arises from the following:

a. **LB**. The number of trees to be considered when computing the optimal solution for a session in isolation is $O\left(\left|s_z \cup \mathbf{D}_z\right| \cdot N^2\right)$, that is the complexity of the TM algorithm; where: s_z is the multicast source; D_z represents the set of destination nodes; and *N* is the number of nodes in the network. This is the first operation in the LB algorithm. Then, $\left|\mathbf{G}_z\right|$ best multicast trees are considered for each session and all the possible combinations of these *M* trees (one for each set $\mathbf{G}_z$) are evaluated; if assuming that querying the cost function (2.1) has time complexity $O(1)$, the overall complexity is: $O\left(M \cdot \left|\mathbf{D}_z\right| \cdot N^2 + \left|\mathbf{G}_z\right|^M\right)$.

b. **GA**: To compute the time complexity of the GA algorithm, some parameters have to be recalled: P_{Dim}, the dimension of the used population; *CROSS*, the crossover percentage; and IT, the number of iterations of the genetic algorithm. As to the first part of the proposed GA-based technique, a set of paths between each multicast source and every destination had to be found out; if considering a not dense graph[1], the time complexity of depth-first search algorithm is $O(N + H)$, where *N* and *H* represent the number of nodes and edges, respectively. Then, applying the depth-first search algorithm for each source-destination pair, a time complexity of: $O(M \cdot (N + H) \cdot \left|\mathbf{D}_z\right|)$

[1] A graph is dense, if the number of edges is close to $N \cdot (N-1)/2$, where *N* is the number of nodes.

was obtained. The subsequent application of the GA in the two phases of the proposed algorithm yields to the following overall time complexity: $O\left(M \cdot (N+H) \cdot |\mathbf{D}_z| + IT \cdot P_{\mathrm{dim}}\right)$.

The resulting time complexity expressions show that the proposed algorithm allows to drastically reduce the required number of operations. In particular, respect to the nodes in the network this number increases linearly in the GA. Differently, in the LB, the time complexity is the maximum among $O\left(N^2\right)$ and $O\left(|\mathbf{G}_z|^M\right)$; in fact, it has to be noted that $|\mathbf{G}_z|$ increases with N as described in the second point. However, the exact relationship between N and $|\mathbf{G}_z|$ was not find out.

Table II.1. Comparison between the lower bound based search (LB) and the proposed genetic algorithms based search method (GA)

Number of sources	LB Cost	GA cost		GA cost reduction		LB processing time	GA processing time	
		P_{dim} =8	P_{dim} =64	P_{dim} =8	P_{dim} =64		P_{dim} =8	P_{dim} =64
1	68	76	68	45.7%	53.4%	1*h* 24*min*	50*sec*	1*min* 18*sec*
5	278	311	287	13.6%	20.2%	4*h* 54*min*	4*min* 32*sec*	6*min* 11*sec*
10	598	679	610	9.5%	18.6%	48*h* 20*min*	9*min* 55*sec*	13*min* 15*sec*

When considering ten sources, a cost reduction of 18.6%, and a final cost quite close to the lower bound were obtained. Table II.1 shows the considered network and the obtained solution in case of five sources: the nodes and links belonging to the final solution are shown with empty squares and dashed lines, respectively. The computational time spent on the last solution (ten sources) was 13*min* and 15*sec*, against 48*h*, 20*min* and 50*sec*.

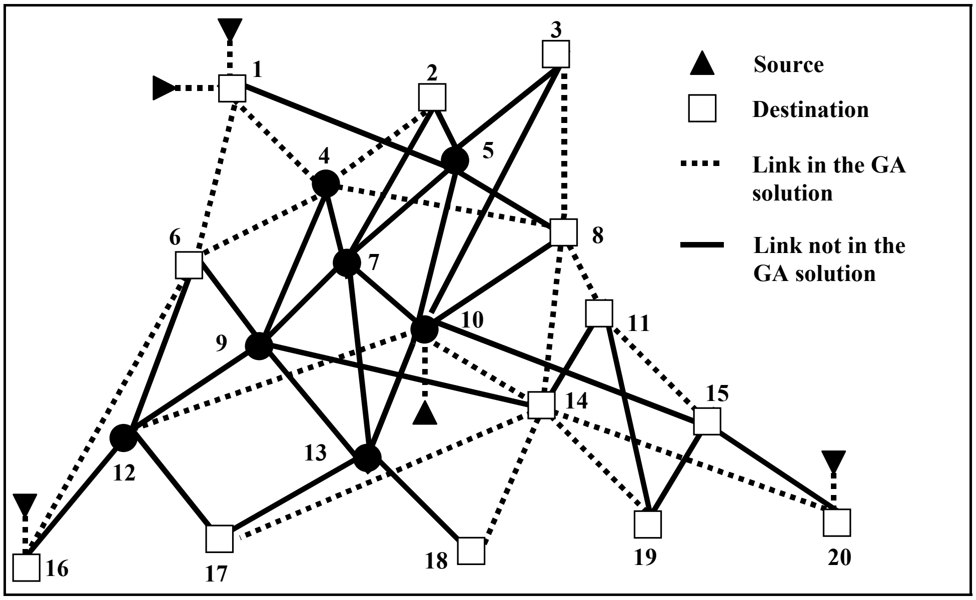

Fig. II.1. Network capacity assignment for LB and GA algorithm

2.2. Power management in Wireless Infrastructure Networks

The major advantage of wireless networks is that network access does not require nodes to be in any particular location. To take full advantage of mobility, nothing can constrain the location of a node, including the availability of electrical power. Mobility therefore implies that most mobile devices can run on batteries. But battery power is a scarce resource; batteries can run only so long before they need to be recharged. Requiring mobile users to return frequently to commercial power is inconvenient, to say the least. Many wireless applications require long battery life without sacrificing network connectivity.

As with any other network interface, powering down the transceiver can lead to great power savings in wireless networks. When the transceiver is off, it is said to be sleeping, dozing, or in power-saving mode (PS). When the transceiver is on, it is said to be awake, active, or simply on. Power conservation in 802.11 is achieved by minimizing the time spent in the latter stage and maximizing the time in the former. However, 802.11 accomplishes this without sacrificing connectivity.

Power management can achieve the greatest savings in infrastructure networks. All traffic for mobile stations must go through access points, so they are an ideal location to buffer traffic. There is no need to work on a distributed buffer system that must be implemented on every station; the bulk of the work is left to the access point. By definition, access points are aware of the location of mobile stations, and a mobile station can communicate its power management state to its access point. Furthermore, access points must remain active at all times; it is assumed that they have access to continuous power. Combining these two facts allows access points to play a key role in power management on infrastructure networks.

Access points have two power management-related tasks. First, because an access point knows the power management state of every station that has associated with it, it can determine whether a frame should be delivered to the wireless network because the station is active or buffered because the station is asleep. But buffering frames alone does not enable mobile stations to pick up the data waiting for them. An access point's second task is to announce periodically which stations have frames waiting for them. The periodic announcement of buffer status also helps to contribute to the power savings in infrastructure networks. Powering up a receiver to listen to the buffer status requires far less power than periodically transmitting polling frames. Stations only need to power up the transmitter to transmit polling frames after being informed that there is a reason to expend the energy.

Power management is designed around the needs of the battery-powered mobile stations. Mobile stations can sleep for extended periods to avoid using the wireless network interface. Part of the association request is the Listen Interval parameter, which is the number of Beacon periods for which the mobile station may choose to sleep. Longer listen intervals require more buffer space on the access point; therefore, the Listen Interval is one of the key parameters used in estimating the resources required to support an association. The Listen Interval is a contract with the access point. In agreeing to buffer any frames while the mobile station is sleeping, the access point agrees to wait for at least the listen interval before discarding frames. If a mobile station fails to check for waiting frames after each listen interval, they may be discarded without notification.

TIM: the Traffic Indicator Map

When frames are buffered, the destination node's AID (association ID) provides the logical link between the frame and its destination. Each AID is logically connected to frames buffered for the mobile station that is assigned that AID. Multicast and broadcast frames are buffered and linked to an AID of zero. Delivery of buffered multicast and broadcast frames is treated in the next subsection.

Buffering is only half the battle. If stations never pick up their buffered frames, saving the frames is a rather pointless exercise. To inform stations that frames are buffered, access points periodically assemble a traffic indication map (TIM) and transmit it in Beacon frames. The TIM is a virtual bitmap composed of 2,008 bits; offsets are used so that the access point needs to transmit only a small portion of the virtual bitmap. This conserves network capacity when only a few stations have buffered data. Each bit in the TIM corresponds to a particular AID; setting the bit indicates that the access point has buffered unicast frames for the station with the AID corresponding to the bit position.

Mobile stations must wake up and enter the active mode to listen for Beacon frames to receive the TIM. By examining the TIM, a station can determine if the access point has buffered traffic on its behalf. To retrieve buffered frames, mobile stations use PS-Poll Control frames. When multiple stations have buffered frames, all stations with buffered data must use the random backoff algorithm before transmitting the PS-Poll.

Each PS-Poll frame is used to retrieve one buffered frame. That frame must be positively acknowledged before it is removed from the buffer. Positive acknowledgment is required to keep a second, retried PS-Poll from acting as an implicit acknowledgment. Fig. II.2 illustrates the process.

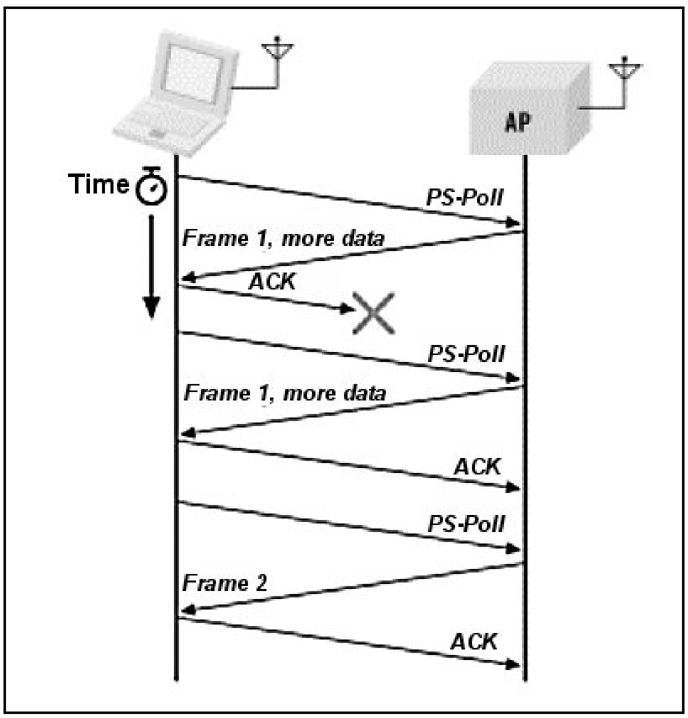

Fig. II.2. PS-Poll frame retrieval

If multiple frames are buffered for a mobile station, then the More Data bit in the Frame Control field is set to 1. Mobile stations can then issue additional PS-Poll requests to the access point until the More Data bit is set to 0, though no time constraint is imposed by the standard.

After transmitting the PS-Poll, a mobile station must remain awake until either the polling transaction has concluded or the bit corresponding to its AID is no longer set in the TIM. The reason for the first case is obvious: the mobile station has successfully polled the access point; part of that transaction was a notification that the mobile station will be returning to a sleeping mode. The second case allows the mobile station to return to a power conservation mode if the access point discards the buffered frame. Once all the traffic buffered for a station is delivered or discarded, the station can resume sleeping.

The buffering and delivery process is illustrated in Fig. II.3 (please note that this figure is somewhat simplified, since a special kind of TIM is used to deliver multicast traffic; it will be described further), which shows the medium as it appears to an access point and two associated power-saving stations. The hash marks on the timeline represent the beacon interval. Every beacon interval, the access point transmits a Beacon frame with a TIM information element 'Station 1' has a listen interval of 2, so it must wake up to receive every other TIM, while 'Station 2' has a listen interval of 3, so it wakes up to process every third TIM. The lines above the station base lines indicate the ramp-up process of the receiver to listen for the TIM.

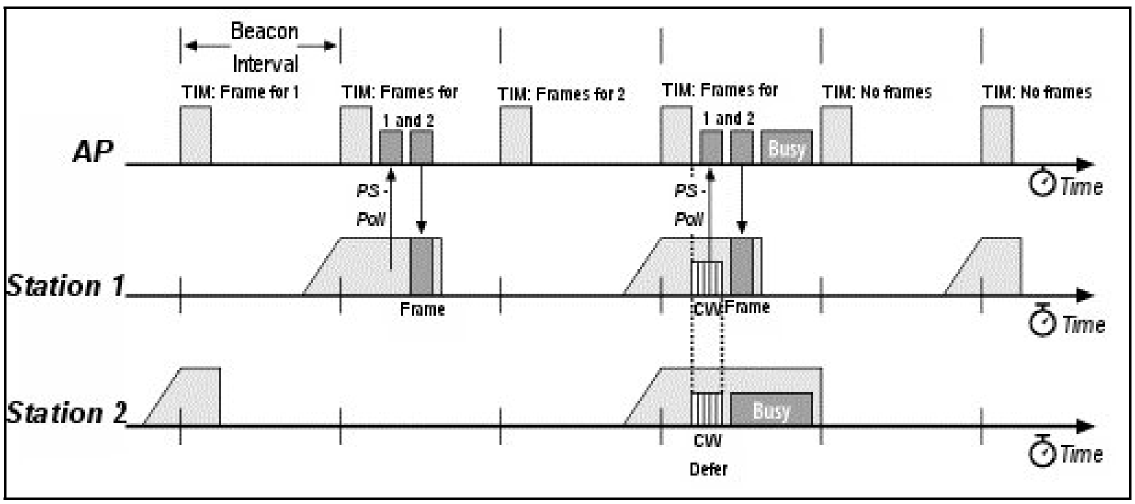

Fig. II.3. Buffered frame retrieval process

At the first beacon interval, there are frames buffered for 'Station 1'. No frames are buffered for 'Station 2', though, so it can immediately return to sleep. At the second beacon interval, the TIM indicates that there are buffered frames for 'Station 1' and 'Station 2', though only 'Station 1' woke up to listen to the TIM. 'Station 1' issues a PS-Poll and receives the frame in response. At the conclusion of the exchange, 'Station 1' returns to sleep. Both stations are asleep during the third beacon. At the fourth beacon, both wake up to listen to the TIM, which indicates that there are frames buffered for both. Both 'Station 1' and 'Station 2' prepare to transmit PS-Poll frames after the expiration of a contention window countdown. 'Station 1' wins because its random delay was shorter. 'Station 1' issues a PS-Poll and receives its buffered frame in response. During the transmission, 'Station 2' defers. If, at the end of that frame transmission, a third station, which is not illustrated, seizes the medium for transmission, ‘Station 2’ must continue to stay awake until the next TIM. If the access point has run out of buffer space and has discarded the buffered frame for ‘Station 2’, the TIM at the fifth beacon indicates that no frames are buffered, and ‘Station 2’ can finally return to a low-power mode.

Stations may switch from a power conservation mode to active mode at any time. It is common for laptop computers to operate with full power to all peripherals when connected to AC power and conserve power only when using the battery. If a mobile station switches to the active mode from a sleeping mode, frames can be transmitted without waiting for a PS-Poll. PS-Poll frames indicate that a power-saving mobile station has temporarily switched to an active mode and is ready to receive a buffered frame. By definition, active stations have transceivers operating continuously. After a switch to active mode, the access point can assume that the receiver is operational, even without receiving explicit notification to that effect.

Access points must retain frames long enough for mobile stations to pick them up, but buffer memory is a finite resource. 802.11 mandates that access points use an aging function to determine when buffered frames are old enough to be discarded. The standard leaves a great deal to the discretion of the developer because it specifies only one constraint. Mobile stations depend on access points to buffer traffic for at least the listen interval specified with the association, and the standard forbids the aging function from discarding frames before the listen interval has elapsed. Beyond that, however, there is a great deal of latitude for vendors to develop different buffer management routines.

DTIM: the Delivery TIM

Frames with a group address cannot be delivered using a polling algorithm because they are, by definition, addressed to a group. Therefore, 802.11 incorporates a mechanism for buffering and delivering broadcast and multicast frames. Buffering is identical to the unicast case, except that frames are buffered whenever any station associated with the access point is sleeping. Buffered broadcast and multicast frames are saved using AID 0. Access points indicate whether any broadcast or multicast frames are buffered by setting the first bit in the TIM to 0; this bit corresponds to AID 0.

Each BSS has a parameter called the DTIM Period. TIMs are transmitted with every Beacon. At a fixed number of Beacon intervals, a special type of TIM, a Delivery Traffic Indication Map (DTIM), is sent. The TIM element in Beacon frames contains a counter that counts down to the next DTIM; this counter is zero in a DTIM frame. Buffered broadcast and multicast traffic is transmitted after a DTIM Beacon. Multiple buffered frames are transmitted in sequence; the More Data bit in the Frame Control field indicates that more frames must be transmitted. Normal channel acquisition rules apply to the transmission of buffered frames. The access point may choose to defer the processing of incoming PS-Poll frames until the frames in the broadcast and multicast transmission buffers have been transmitted.

Fig. II.4 shows an access point and one associated station. The DTIM interval of the access point is set to 3, so every third TIM is a DTIM. 'Station 1' is operating in a sleep mode with a listen interval of 3. It will wake up on every third beacon to receive buffered broadcast and multicast frames. After a DTIM frame is transmitted, the buffered broadcast and multicast frames are transmitted, followed by any PS-Poll exchanges with associated stations. At the second beacon interval, only broadcast and multicast frames are present in the buffer, and they are transmitted to the BSS. At the fifth beacon interval, a frame has also been buffered for 'Station 1'. It can monitor the map in the DTIM and send a PS-Poll after the transmission of buffered broadcast and multicast frames has concluded.

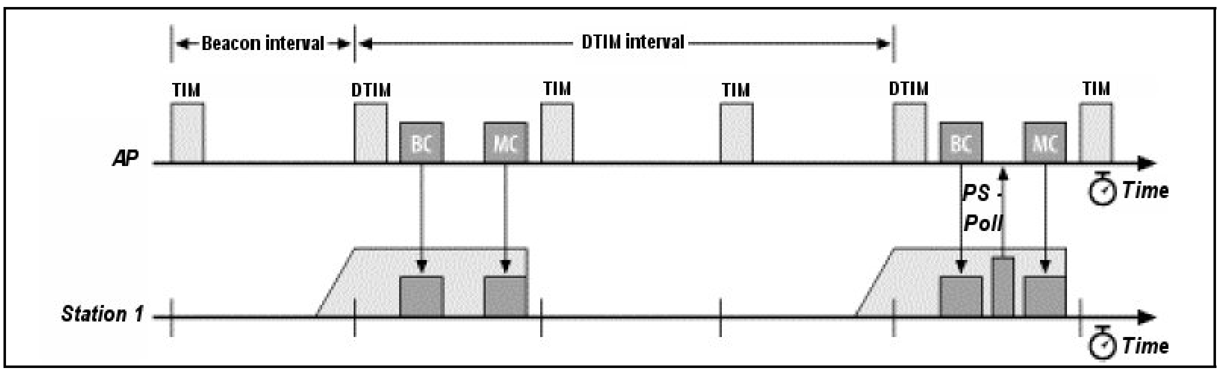

Fig. II.4. Multicast and broadcast buffer transmission after DTIMs

To receive broadcast and multicast frames, a mobile station must be awake for DTIM transmissions. Nothing in the specification, however, keeps power-saving stations in infrastructure networks from waking up to listen to DTIM frames. Some products that implement power-saving modes will attempt to align their awakenings with DTIM transmissions. If the system administrator determines that battery life is more important than receiving broadcast and multicast frames, a station can be configured to sleep for its listen period without regard to DTIM transmissions. Some documentation may refer to this as extremely low power, ultra power-saving mode, deep sleep, or something similar.

Several products allow configuration of the DTIM interval. Lengthening the DTIM interval allows mobile stations to sleep for longer periods and maximizes battery life at the expense of timely delivery. Shorter DTIM intervals emphasize quick delivery at the expense of more frequent power-up and power-down cycles. You can use a longer DTIM when battery life is at a premium and delivery of broadcast and multicast frames is not important. Whether this is appropriate depends on the applications you are using and how they react to long link-layer delays.

2.3. QoS management in 3G Netwroks

QoS high level requirements

In [3G231] a set of requirements at high level for the Quality of Service are provided. These are:

- end user QoS requirements,
- general QoS requirements,
- technical requirements.

End user QoS requirements

Generally, end users care only the issues that are visible to them. The involvement of the user leads to the following conclusions. From the end-user point of view:

- only the QoS perceived by end-user matter;
- the number of user defined/controlled attributes has to be as small as possible;
- derivation/definition of QoS attributes from the application requirements has to be simple;
- QoS attributes shall be able to support all applications that are used, a certain number of applications have the characteristic of asymmetric nature between two directions, uplink/downlink;
- QoS definitions have to be future proof;
- QoS has to be provided end-to-end.

General requirements for QoS

- QoS attributes (or mapping of them) should not be restricted to one or few external QoS control mechanisms but the QoS concept should be capable of providing different levels of QoS by using UMTS specific control mechanisms (not related to QoS mechanisms in the external networks).
- All attributes have to have unambiguous meaning.
- QoS mechanism have to allow efficient use of radio capacity.
- Allow independent evolution of Core and Access networks.

- Allow evolution of UMTS network, (i.e., eliminate or minimise the impact of evolution of transport technologies in the wireline world).
- All attribute combinations have to have unambiguous meaning.

Technical requirements for QoS

This clause presents the general high-level technical requirements for the UMTS QoS. QoS will be defined with a set of attributes. These attributes should meet the following criteria:

- UMTS QoS control mechanisms shall provide QoS attribute control on a peer to peer basis between UE and 3G gateway node;
- the UMTS QoS mechanisms shall provide a mapping between application requirements and UMTS services;
- the UMTS QoS control mechanisms shall be able to efficiently interwork with current QoS schemes. Further, the QoS concept should be capable of providing different levels of QoS by using UMTS specific control mechanisms (not related to QoS mechanisms in the external networks);
- a session based approach needs to be adopted for all packet mode communication within the 3G serving node with which UMTS QoS approach shall be intimately linked, essential features are multiple QoS streams per address;
- the UMTS shall provide a finite set of QoS definitions;
- the overhead and additional complexity caused by the QoS scheme should be kept reasonably low, as well as the amount of state information transmitted and stored in the network;
- QoS shall support efficient resource utilisation;
- the QoS attributes are needed to support asymmetric bearers;
- applications (or special software in UE or 3G gateway node) should be able to indicate QoS values for their data transmissions;
- QoS behaviour should be dynamic , i.e., it shall be possible to modify QoS attributes during an active session;
- number of attributes should be kept reasonably low (increasing number of attributes, increase system complexity);
- user QoS requirements shall be satisfied by the system, including when change of SGSN within the Core Network occurs.

QoS Architecture

Network Services are considered end-to-end, this means from a Terminal Equipment (TE) to another TE. An End-to-End Service may have a certain Quality of Service (QoS) which is provided for the user of a network service. It is the user that decides whether he is satisfied with the provided QoS or not.

An end-to-end service could be considered, following the indication in [3G231], as composed by many blocks, i.e. services, involved in different UMTS architecture levels.

To realise a certain network QoS a Bearer Service with clearly defined characteristics and functionality is to be set up from the source to the destination of a service.

A bearer service includes all aspects to enable the provision of a contracted QoS. These aspects are among others the control signalling, user plane transport and QoS management functionality. A UMTS bearer service layered architecture is depicted in Fig. II.5, each bearer service on a specific layer offers its individual services using services provided by the layers below.

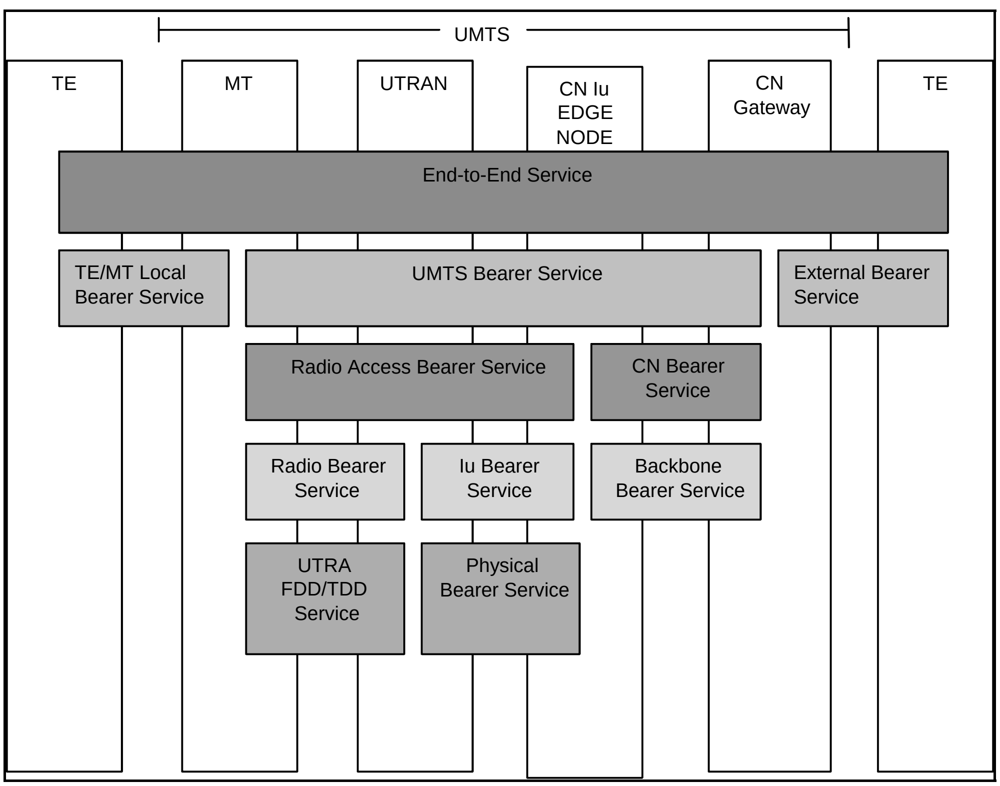

Fig. II.5. UMTS QoS Architecture

The End-to-End Service and UMTS Bearer Service

On its way from the TE to another TE the traffic has to pass different bearer services of the network(s). A TE is connected to the UMTS network by use of a Mobile Termination (MT). The End-to-End Service on the application level uses the bearer services of the underlying network(s). The *End-to-End-Service* used by the TE will be realised using a *TE/MT Local Bearer* Service, a UMTS Bearer Service, and an External Bearer Service. *TE/MT Local Bearer Service* is not further elaborated here as this bearer service is outside the scope of the UMTS network. Since UMTS operator offers the various services offered by the *UMTS Bearer Service*, it is this bearer service that provides the UMTS QoS. The *External Bearer Service* is not further elaborated here as this bearer may be using several network services, e.g. another *UMTS Bearer Service*. Other services under UMTS bearer service are not considered since out of the scope of this research.

UMTS QoS Classes

When defining the UMTS QoS classes, also referred to as traffic classes, the restrictions and limitations of the air interface have to be taken into account. It is not reasonable to define complex mechanisms as have been in fixed networks due to different error characteristics of the air interface. The QoS mechanisms provided in the cellular network have to be robust and capable of providing reasonable QoS resolution. Table II.2 illustrates the QoS classes for UMTS.

There are four different QoS classes as stated in [3G231]:

- Conversational class;
- Streaming class;
- Interactive class; and
- Background class.

The main distinguishing factor between these QoS classes is how delay sensitive the traffic is: Conversational class is meant for traffic which is very delay sensitive while Background class is the most delay insensitive traffic class.

Conversational and Streaming classes are mainly intended to be used to carry real-time traffic flows. The main divider between them is how delay sensitive the traffic is. Conversational real-time services, like video telephony, are the most delay sensitive applications and those data streams should be carried in Conversational class.

Interactive class and Background are mainly meant to be used by traditional Internet applications like Web browsing, Email, Telnet, FTP (File Transfer Protocol) and News. Due to looser delay requirements, compare to conversational and streaming classes, both provide better error rate by means of channel coding and retransmission. The main difference between Interactive and Background class is that Interactive class is mainly used by interactive applications, e.g. interactive Email or interactive Web browsing, while Background class is meant for background traffic, e.g. background download of Emails or background file downloading. Responsiveness of the interactive applications is ensured by separating interactive and background applications. Traffic in the Interactive class has higher priority in scheduling than Background class traffic, so background applications use transmission resources only when interactive applications do not need them. This is very important in wireless environment where the bandwidth is low compared to fixed networks.

Table II.2. UMTS QoS classes

Traffic class	Conversational class conversational RT	Streaming class streaming RT	Interactive class Interactive best effort	Background Background best effort
Fundamental characteristics	- Preserve time relation (variation) between information entities of the stream Conversational pattern (stringent and low delay)	- Preserve time relation (variation) between information entities of the stream	- Request response pattern - Preserve payload content	- Destination is not expecting the data within a certain time - Preserve payload content
Example of the application	- voice	- streaming video	- Web browsing	- background download of emails

UMTS QoS Attributes

UMTS bearer service attributes describe the service provided by the UMTS network to the user of the UMTS bearer service. A set of QoS attributes (QoS profile) specifies this service. At UMTS bearer service establishment or modification different QoS profiles have to be taken into account.

- The UE capabilities form a QoS profile which may limit the UMTS bearer service which can be provided.
- The UE or the terminal equipment (TE) within the terminating network may request a QoS profile at UMTS bearer establishment or modification. The application using the UE may request the UE to provide a UMTS bearer service with a specific QoS profile. If the application requests no specific QoS the UE may use a QoS profile configured within the UE (e.g., by AT commands). How the TE derives a QoS profile is out of scope for UMTS.
- A QoS profile in the UMTS subscription describes the upper limits for the provided service if the service user requests specific values.

- If the UE requests or modifies a UMTS bearer and one or more of the QoS attributes are not specified by the UE by setting the attributes to 'subscribed', the SGSN shall assume a request of values as specified in the QoS profile in the UMTS subscription. If the UE sets the traffic class to 'subscribed', the SGSN shall assume a request for Interactive class. When the application in the UE requires streaming or conversational QoS, then the UE shall at least explicitly request the traffic class and should explicitly request the guaranteed bit rate and the maximum bit rate. For the rest of the QoS attributes, the network shall ensure that the negotiated QoS contains only values explicitly defined for the traffic class.
- A Network specific QoS profile characterising for example the current resource availability or other network capabilities or limitations may limit the provided UMTS bearer service or initiate a modification of an established UMTS bearer service.

Table II.3. UMTS QoS attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/Retention priority	X	X	X	X
Source statistics descriptor	X	X		
Signalling indication			X	

The definition of the QoS attributes contained in the table and their respective values are not shown here, but they can be retrieved in [3G231].

When establishing a UMTS bearer and the underlaying Radio Access Bearer for support of a service request, some attribute on UMTS level does typically not have the same value as corresponding attribute on Radio Access Bearer level. For example requested transfer delay for the UMTS bearer shall typically be larger that the requested transfer delay for the Radio Access Bearer, as the transport through the core network will use a part of the acceptable delay.

Application characteristics

Once defined the concept of end user QoS, the architecture of end-to-end services, and the UMTS QoS classes it is necessary to define the applications which can be considered, their characteristics, and how these services can be mapped into the UMTS QoS classes.

Fig. II.6 below, summarises the major groups of application in terms of QoS requirements[see also 3G221]. Traditional and new applications may be applicable to one or more groups. However, there is no strict one-to-one mapping between the groups of application/service defined here and the traffic classes as previous described. For instance, an Interactive application/service can very well use a bearer of the Conversational traffic class if the application/service or the user has tight requirements on delay.

Error tolerant	Conversational voice and video	Streaming audio and video	Voice messaging	Fax
Error intolerant	Telnet, interactive games	Still image, paging	E-commerce, WWW browsing,	E-mail arrival notification
	Conversational application	Streaming application	Interactive application	Background application

Fig. II.6. Summary of applications in terms of error tolerance

All these applications can be provided over a Circuit Switched or Packet Switched Core Network domain.

Conversational real-time applications

The most well known use of this scheme is telephony speech (e.g. GSM), but with Internet and multimedia a number of new applications will require this scheme, for example voice over IP and video conferencing tools. Real time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception (the senses). Therefore this scheme raises the strongest and most stringent QoS requirements. Applications which will be considered are:

- conversational voice,

- videophone,
- telemetry- two-way control,
- interactive games,
- telnet.

Streaming applications

When the user is looking at (listening to) video (audio) the scheme streams applies. The real time data flow is always aiming at a live (human) destination. It is a one way transport. Applications which will be considered are:

- speech, mixed speech and music, medium and high quality music;
- movie clips, surveillance, real-time video;
- bulk data transfer/retrieval, layout and synchronisation information;
- still image.

Interactive applications

When the end-user, that is either a machine or a human, is on line requesting data from remote equipment (e.g. a server), this scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, server access. Examples of machines interaction with remote equipment are: polling for measurement records and automatic data base enquiries (tele-machines). Applications which will be considered are:

- voice messaging;
- MMS/SMS (Multimedia Messaging Service/ Short Message Service);
- web-browsing – HTML (HyperText Markup Language);
- transaction services - high priority e.g. e-commerce, ATM;
- e-mail (server access).

Background applications

When the end-user, that typically is a computer, sends and receives data-files in the background, this scheme applies. Examples are background delivery of e-mails, SMS, download of databases and reception of measurement records. Applications which will be considered are:

- fax;
- e-mail arrival notification;
- low priority transaction service.

Chapter III

QoS optimization for multicast services

Due to the growth of multimedia services, the bandwidth is becoming even more important; for this reason many multimedia real time applications use multicast communication (see Chapter I, Section 1.1) to conserve network bandwidth. First research works dealt with a single multicast session and focused on minimizing the transmission cost of the single routing tree. Many heuristic algorithms ([SLS96]- [TM80]) have been presented to solve the NP-complete unconstrained case, known as the Steiner tree problem. Other works as ([J95], [ZPG95], [CHH97]) have extended this problem by introducing the constraint on quality of service (QoS) requirements, such as end-to-end transmission delay.

Since in the real world several multicast sessions occur simultaneously, a new and more complex optimization problem needed to be represented: the group multicast routing problem. Until now, only few related papers have been published ([WLJ02], [PAS95], [CGY00], [WLJ02]). In particular Chen *et al.* [CGY00] used an integer-programming approach considering only sessions with the same bandwidth requirements, while Wang *et al.* [WLJ02], proposed an heuristic approach using a simple distance based cost function.

In particular, Chen *et al.* [CGY00] defined the multicast packing problem in which the network tried to accommodate simultaneously all the multicast groups while avoiding bottlenecks on the links with high throughput (i.e., minimized the maximum congested link shared among multicast groups). Minimization of maximum congestion is achieved at the expense of increasing the size of some multicast trees which in turn affected the delay. This trade-off was addressed by adding a penalty term to the objective function of the optimal packing formulation. The penalty term was a function of the amount of dilation from the size of the optimal tree obtained for each multicast session independently from the others (i.e., in isolation). Since the mathematical programming formulation for the optimization problem was computationally intractable, they resorted to suboptimal solutions with heuristics. Their heuristic method aimed to reduce the sharing of a link while ensuring that the size of multicast trees will never exceed alpha times the size of the optimum tree for a multicast group in isolation. Optimum multicast tree for each group (in isolation) was computed by using cutting-plane inequalities and the branch-and-cut algorithm.

Differently, Wang *et al.* [WLJ02], considered a multicast routing problem with multiple multicast sessions under a capacity limited constraint (there is no analysis on delay). In order to solve this problem they proposed two heuristic algorithms, Steiner-tree-based and cut-set-based. If the available bandwidth for the service is just enough, these algorithms may fail to find a solution even if the solution exists.

On the other hand, the genetic approach is gaining an increasing interest for solving complex problems in the networking thematic, as network design [AR04] [Chapter II, Section 2.1] and simple routing [AR02]. GA for multicast routing without constraints was presented by Hwang *et al.* [HDY00] and Bhattacharya *et al.* [BVSN05], while Chen *et al.* [CYX04] and Hamdan *et al.* [CYX04] treated the QoS provided for real-time applications for a single multicast session, too.

In this Chapter the problem of group multicast routing is addressed, and an innovative approach proposed. One of the novel features of the proposed solution consists of using the GA to reduce the computation complexity so that the proposed algorithm can be applied in a real-time context. Additionally, it has strived to define a cost function $f_C(D,B)$ that weights delay (*D*) and network resource utilization (*B*) components together. This has been defined by identifying some combinations of these two components that are advantageous, disadvantageous, or optimal from the operator point of view. Regions encompassing these combinations have been defined in the delay-bandwidth utilization plane (called *D-B* plane). The exact regions border definition is left to the specific operator needs that may vary from case to case. To this, the cost function expression has been defined in terms of few basic parameters.

The Chapter is organized as follows. Section 3.1 describes the problem of group multicast routing. Section 3.2 illustrates the proposed GA-based solution. Section 3.3 analyses the overall complexity, while experimental results are shown in the last section.

3.1. Group multicast routing problem

The problem of group multicast routing is defined as follows: given an existing network with known unicast traffic, find the optimal link capacity assignment to accommodate the multicast traffic generated by a group of multicast sources. The optimality has to be defined on the basis of the type of services conveyed through the multicast sessions and the operator objectives; yet, bandwidth usage and transmission delay are widely used in this context.

The communication network is modelled by a directed graph $G(\mathbf{V},\mathbf{E})$, where $\mathbf{V}$ is a finite set of vertices (network nodes) n_i and $\mathbf{E}$ is the set of edges (network links) l_{ij} with capacity c_{ij} and background traffic b_{ij}. The number of nodes and links (i.e., the cardinalities of $\mathbf{V}$ and $\mathbf{E}$) are *N* and *E*, respectively. Let $\mathbf{S}$ be a subset of $\mathbf{V}$ denoting the group of multicast source nodes: $\mathbf{S}\subset\mathbf{V}$, $|\mathbf{S}|=S$, and $S<N$. For each session z, a source node s_z transmits to its multicast destination group $\mathbf{M}_z$ at rate y_z. The set $\mathbf{M}_z$ is included in V and the number of m_{kz} nodes in $\mathbf{M}_z$ is K_z, with $z = 1,...,S$. A solution to the group multicast routing problem was represented by a set of multicast trees $\mathbf{F} = \{\mathbf{T}_1,...,\mathbf{T}_S\}$, where each tree $\mathbf{T}_z$ is rooted at s_z and $\mathbf{M}_z$ is the leaf nodes set. There may be many solutions for a set of multicast sessions. The optimal one is represented by the set of trees $\mathbf{F}_{opt}$ that minimizes a certain cost function under a given set of constraints. Solving the group multicast routing problem is equivalent to find the optimal set of distribution trees on the basis of such cost function, which usually encompasses network usage and transmission delay.

3.2. Proposed solution

In general, a genetic algorithm has five basic components as follows:

1. An encoding method, which is a genetic representation (genotype) of solutions to the program.
2. A way to create an initial population of individuals (chromosomes).
3. An evaluation function, rating solutions in terms of their fitness, and a selection mechanism.
4. the genetic operators (crossover and mutation) that alter the genetic composition of offspring during reproduction
5. Values for the parameters of genetic algorithm.

Since the solution relies on using a genetic algorithm, this proposal was described covering these five topics in the following subsections.

Encoding method

At high level, each group of multicast trees was represented as the overlapping of these multicast session trees; each tree, at its turn, was represented as the overlapping of all the source-destination unicast paths composing the tree.

Since the chromosome had to represent a possible solution of the group multicast routing problem, the chromosome is created by combining S genes, each one coding a specific multicast tree: the first gene coded the first session, and so on. Each gene g_z is in fact an integer number which is the row index of the matrix containing all the possible multicast trees for the z-th session. The row of this matrix is a vector $\mathbf{L}_z$ which codes a specific multicast tree.

Because each tree is composed by different unicast source-destination paths (where the source is always the same), then each element of the vector $\mathbf{L}_z$ codes a specific path: the first element codes the path from the source to the first destination, and so on. The length of the vector is thus equal to K_z, i.e. the number of destinations. Referring to a tree coded by $\mathbf{L}_z$, all the possible unicast paths from the source s_z to a specific destination m_{kz} are stored in a $\mathbf{L}_{kz}$ matrix; Then, each element of $\mathbf{L}_z$ is the row index of its $\mathbf{L}_{kz}$ matrix.

The rows of $\mathbf{L}_{kz}$ matrix are unicast paths represented as a list of nodes along the path: the first element is the source while the last one is the destination. The value of these elements is the *i*-th index of the n_i node.

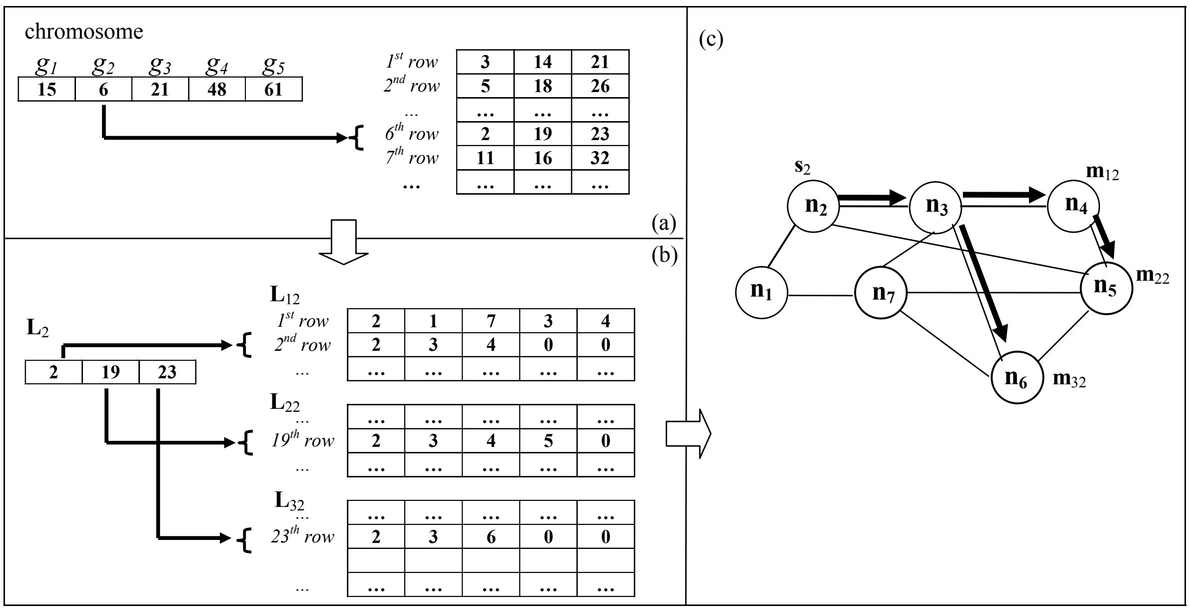

Fig. III.1. Genotype coding

For a better understanding, a simple example is provided in Fig. III.1. Just suppose that in the network five multicast flows occur, then the chromosome will be composed of five genes, each one representing a particular multicast tree. As respect to Fig. III.1 (a) and (b), the second gene, g_2, is referred to the second session (z = 2) and its value is six, i.e. it refers to the sixth row of the possible multicast tree for this session ($\mathbf{L}_2$); each tree is represented as an array whose length is equal to the number of destinations K_z (in this case K_2 is equal to three) and its values are indexes referred to a particular unicast path source-destination, i.e. each tree is represented as a union of different paths with the same source while different destinations: as respect to Fig. III.1 (b) and (c), the multicast tree coded by $\mathbf{L}_2$ is provided by the paths n_2 - n_3 - n_4, n_2 - n_3 - n_4 - n_5 and n_2 - n_3 - n_6.

Besides the genotype coding, a method for building the related space solution has been proposed. In fact, as every GA-based algorithm, the space solution (which will be further coded in genes and chromosomes) need to be built before running the algorithm, i.e. the solution will be searched in a known finite space. This space is represented by the possible multicast trees combination. Since each tree has been expressed as the union of paths with the same source while different destinations, it is needed at first to elaborate these possible paths. Once they were obtained, all the $\mathbf{L}_z$ arrays were created by permutation and ordered so as to be indexed by a scalar g_z. The resulting number of $\mathbf{L}_z$ arrays had been experimentally determined to be lower than $H_N \cdot \ln N$, where H_N is the maximum out-degree of the network nodes.

In order to reduce memory allocation, the paths are not created by means of the DFS (Depth First Search) algorithm, differently from other GA-algorithm proposed in network literature ([AR02], [HDY00], [BVSN05], [CYX04], [HE04], [HFDMG04]). Therefore a different approach is proposed: the shortest paths and some additional paths are searched by means of Dijkstra algorithm and an heuristic procedure. These steps are fully described below.

Dijkstra paths computation

First of all, the paths with the lowest number of hops has to be discovered by applying the Dijkstra shortest path algorithm between each source-destination couple with a unitary link cost. Since a low number of hops means low end-to-end delays and low number of link usage, these paths represented a good starting point for creating other paths, because a space solution relevant as almost as possible need to be created.

Additional path computation

Since some other paths need to be searched for, it was decided to compute the Dijkstra path not only for each source-destination couple, but for every couple of nodes, even if it not correspond to a source-destination pair in any multicast session and without caring of c_{ij}, b_{ij} values or any other parameter. Then, for each source s_z and each destination m_{kz}, the additional paths were discovered starting from the related shortest path. For this scope it was used a heuristic procedure based on analyzing the nodes adjacent to those belonging to the shortest path. The algorithm is shown in Fig. III.2 and has to be repeated for each couple (s_z, m_{kz}). Let Dij(n_i, n_j) be the shortest path (found in the previous step) from n_i to n_j.

```
Algorithm SearchAdditionalPaths;
    Input: Dij(n_i, n_j) for i,j = 1, ..., N; s_z ; m_kz .
    Output: p additional paths (AP_p).
    begin
        l = number of the nodes in Dij( s_z , m_kz );
        SP( s_z , np_l) = Dij( s_z , m_kz ),
        p = 1;
        while (l > 1) do
          begin
            l = l - 1;
            NA ≡ set of na_x nodes adjacent to np_l for x = 1, ..., |NA|;
            for x = 1 to |NA| do
              begin
                if na_x ∉ Dij( s_z , m_kz ) then
                  begin
                    AP_p = SP( s_z , np_l) ∪ (np_l , na_x) ∪ Dij(na_x , m_kz );
                    p = p+1;
                  end;
              end;
```

Fig. III.2. The algorithm used for searching additional path starting from a shortest one

For a better comprehension, a simple example is presented in Fig. III.3 where the node n_1 is the source s_z of a generic z-th session with n_4 and n_5 as destinations (m_{1z} and m_{2z}). In (a) the Dijkstra path from the destination node n_4 is represented by the nodes and links with straight lines, i.e. n_1 - n_2 - n_3 - n_4, path. Note that there are two destinations for this multicast session, n_4 and n_5, but it has to be considered one source-destination path at a time (n_1 to n_4 in this example). Additional paths were found by considering the node before the last one, and searching for a new Dijkstra path as showed in (b) and (c); this procedure, as also indicated in Fig. III.3, is repeated for all the nodes belonging to the path till the first node, as shown in (d) where the path $n_1 – n_7 – n_5$ - n_4 is found.

Once these additional paths for each source-destination couple were found, the $\mathbf{L}_{kz}$ arrays representing the set of paths for every source-destination pair are built, inserting the Dijkstra source-destination paths and the additional paths.

This procedure then did not create a complete space, but a relevant subspace which does not prevent to obtain good solutions; in fact, since many alternative paths (depending to the output degree of the nodes) are considered, and the chromosome is a combination of path combinations, it is thus possible to build a relevant space. Furthermore, as proved in the last Section, when comparing the results obtained with this technique with the ones obtained by using the DFS, similar results were found.

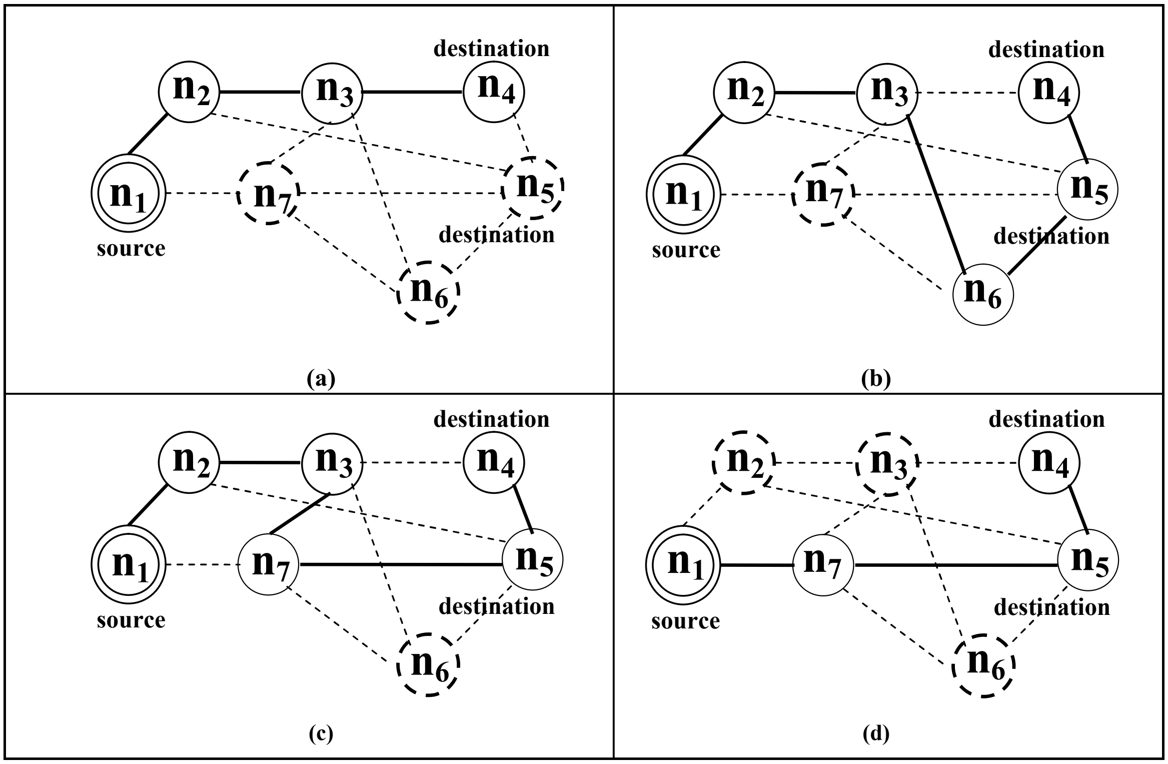

Fig. III.3. Additional paths

Initialization

As usually done in GA literature (see Chapter I, Section 1.4), the population is randomly initialized, and, referring to the general structure of a GA in Fig. I.10, a population P of P_{Dim} chromosomes is created, each composed by S genes.

Evaluation

The chromosome fitness was evaluated by either f_B or f_C. The first was used when the optimality is correlated with a uniform distribution of the traffic over the entire network. The second was used when the optimality depends on both network resource occupation and transmission delay. The exact definition of these cost functions is provided in the following sub-sections.

Bandwidth contribution

The proposed cost function f_B was intended to evaluate the bandwidth distribution over the network links so as to obtain the most uniform resource utilization. In this way, broad trees with low average link utilization could be selected even if there are smaller trees, but with higher average link utilization. Additionally, a constraint on the overall transmission delay for each path in each multicast session was enforced.

For a given solution **F**, let B represent the bandwidth contribution value computed as the average link occupation:

$$B = \frac{1}{L}\sum_{i=1}^{N}\sum_{j=1}^{N}\frac{A(y_{ij},c_{ij})}{c_{ij}}, \tag{3.1}$$

where y_{ij} represents the bandwidth allocation obtained summing the multicast transmission rate to the background traffic occupancy in the l_{ij} link:

$$y_{ij} = b_{ij} + \sum_{z=1}^{S} x_{zij} \cdot y_z \text{ , with } \begin{cases} x_{zij} = 1 & \text{if } l_{ij} \in \mathbf{T}_z \\ x_{zij} = 0 & \text{otherwise} \end{cases}. \tag{3.2}$$

If this allocation exceeds a threshold value B_{Max} (default value equal to 90%), the link is treated as in congestion state:

$$A(a,b) = \begin{cases} a & \text{if } a < 0.9 \cdot b \\ b & \text{otherwise} \end{cases}. \tag{3.3}$$

L is the number of links involved in session routing:

$$L = \sum_{i=1}^{N}\sum_{j=1}^{N} x_{ij} \text{ ,with } \begin{cases} x_{ij} = 1 & \text{if } l_{ij} \subset \mathbf{F} \\ x_{ij} = 0 & \text{otherwise} \end{cases}. \tag{3.4}$$

The cost function f_B proposed is exactly equal to the expression of B in (3.1).

The delay constraint is the following:

$$d_{kz} \leq D_{MAX} , \tag{3.5}$$

that is, for each unicast path the delay has to be less than the threshold D_{MAX} (e.g., 200 msec is commonly used value for streaming applications). The one-way delay d_{kz} from source z to destination m_{kz} is computed as the sum of delays d_{ij} in each hop n_i-n_j involved in the unicast path. The link delay computation used is proposed in [ZHJL03]:

$$d_{ij} = \frac{MTU}{c_{ij} - b_{ij}} + \frac{MTU}{c_{ij}} + \pi_{ij} . \tag{3.6}$$

It is based on the sum of three terms: the expected best packet scheduling delay using Weighted Fair Queuing discipline with a maximum packet length equal to the *MTU*; the packet transmission time through the output link; and the link propagation delay π_{ij}.

Bandwidth and delay contribution: D-B plane

The aim of the second fitness function f_C was to weight both bandwidth usage and one-way transmission delay. These two components were combined together in a cost function that allowed the operator to obtain the “optimal” compromise between service delay and network resource usage. Clearly, there is not a universal definition of optimality in this framework because this depends on the network operator point of view. Accordingly, herein it is defined a parametric cost function that provided a simple way to combine delay and network usage by setting a few parameters.

The bandwidth component used in cost function is the same used in (3.1). As to the delay component, the average value among the multicast tree delays is computed:

$$D = \left(\sum_{z=1}^{S} \alpha_z \cdot D_z \right) \Big/ \left(D_{MAX} \cdot \sum_{z=1}^{S} \alpha_z \right), \tag{3.7}$$

where the multicast tree delay D_z is calculated as the average delay among the unicast paths that constitute the tree for the session z:

$$D_z = \frac{1}{K_z} \cdot \sum_{k=1}^{K_z} d_{kz} \,. \tag{3.8}$$

In order to differentiate real-time applications from simple data transfer ones, weights α_z are introduced.

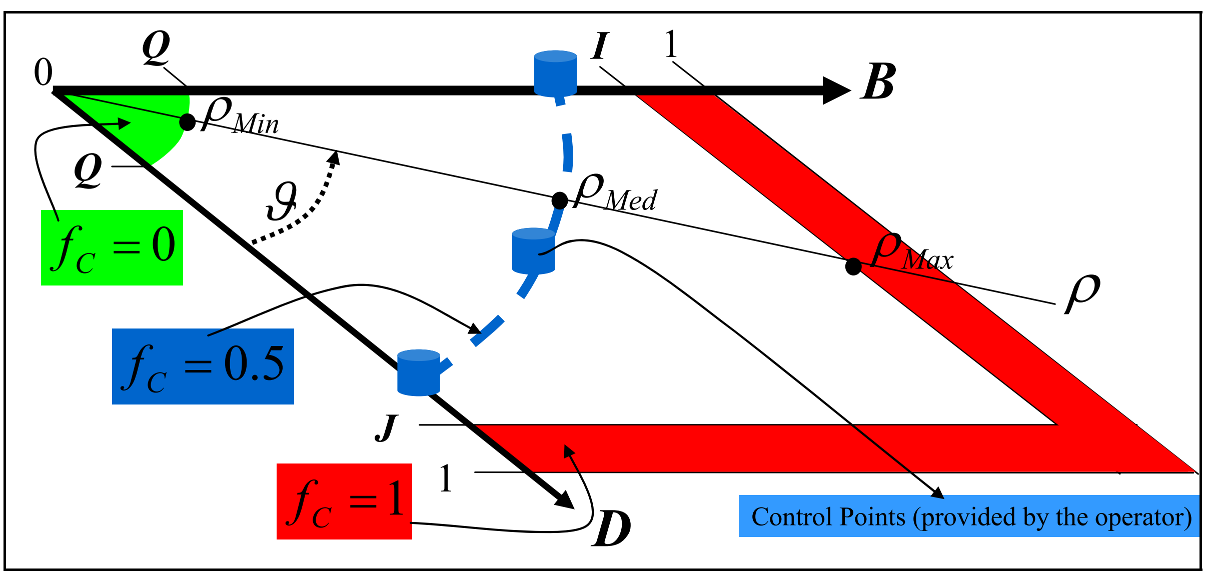

Fig. III.4. *D-B* plane: green area represents the good region; red area is the bad region; and dashed blue line is the control curve

As previously mentioned, the transmission delay and the bandwidth usage were combined in a single cost function $f_C(D, B)$ that provides the goodness of each solution. Such a function is described by defining some particular areas in the bi-dimensional *D-B* plane. Note that both *D* and

B are normalized in the range 0÷1. These areas correspond to combinations of delay and capacity usage that are either particularly advantageous, disadvantageous, or quite common for the network operator. The first encompasses low D and B values and represents situations of very low network resource utilization and low delays. It is highlighted in green colour in Fig. III.4 and $f_C = 0$ is assigned for every point within. It is named *good region* in the following for presentation convenience. The opposite situation corresponds to the “border” area of the *D*-*B* plane, where the capacity usage reaches high values and/or the delay is very high. This is the red area in Fig. III.4, named *bad region*, to which a value of $f_C = 1$ is assigned. Besides these two areas, it was defined a series of combinations of D and B that represent the most frequent operational points for the network and are considered of equal importance for the operator: f_C is set to 0.5 in these points; these combinations are defined by means of a parabolic curve, named *control curve*, determined by the operator defining three points (in terms of *D*-*B* coordinates values); these *control points* are represented by the blue cylindrical symbols in Fig. III.4. The defined three areas constrain the shape of $f_C(D,B)$.

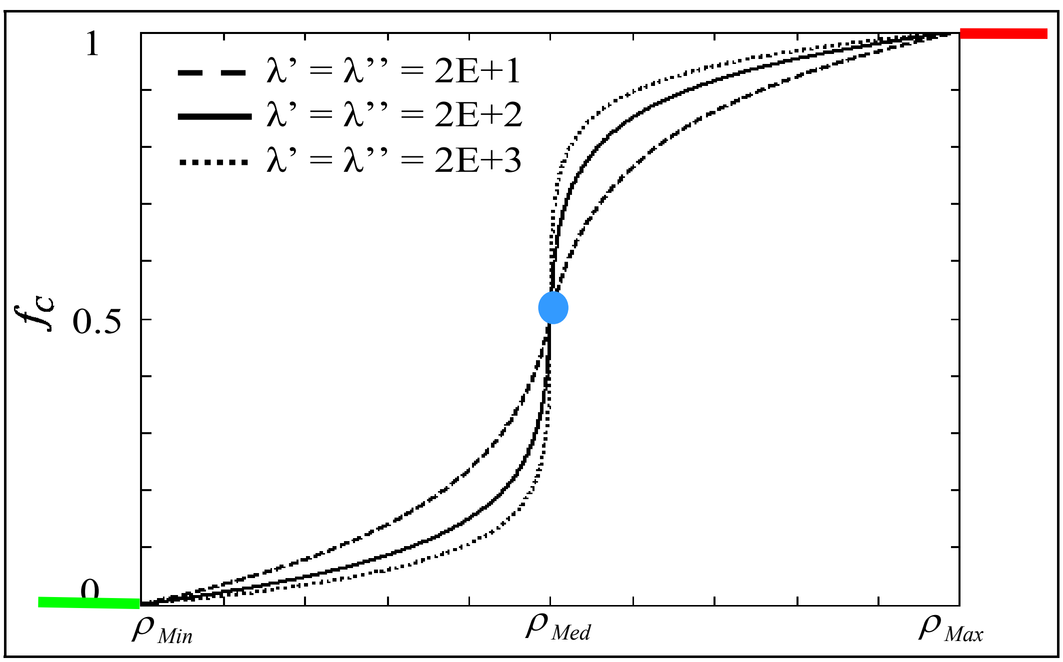

Fig. III.5. Cost function behaviour for a given angle for three different values of λ' and λ''

Additionally, it is reasonable to suppose the cost function convexly increases from the good region border to the control curve and concavely increases from the control curve to the bad region border. Three curves characterized by such a behaviour are depicted in Fig. III.5. In particular, these represent potential curves obtained from the intersection of the bi-dimensional cost function with a plane orthogonal to the *D*-*B* plane and crossing its origin.

Once outlined the main characteristics of the cost function, its analytical expression need to be defined. In particular, it has to be characterized by some parameters to be set by the operator and related to the previously introduced areas. In the following, a possible solution is proposed.

As to the control curve, it is given by the following expression:

$$B = \alpha D^2 + \beta D + \chi, \tag{3.9}$$

where α, β, and χ are easily determined by enforcing the curve to cross the three control points.

The good and bad areas are defined by means of the following inequalities:

bad area: $D \geq J, \forall B; \quad B \geq I, \forall D$,

good area: $D \leq Q, \quad B \leq Q \sin(\arccos D/Q)$,

where *J* and *I* are two parameters defined by the operator and depicted in Fig. III.4.

In order to simplify the tractability, let's now move to the polar representation of the *D*-*B* plane, in terms of the angle ϑ and radius ρ:

$$\vartheta = \arctan B/D; \quad \rho = \sqrt{B^2 + D^2}. \tag{3.10}$$

At a given angle ϑ, the function is desired to have the shape drawn in Fig. III.5 as a function of the radius. With reference to this figure, ρ_{Med} represents the value of ρ on the control curve, ρ_{Min} the value of ρ on the good region border, and ρ_{Max} the value of ρ in the bad region border. Accordingly, for a given value of the angle, the function has the following expression:

$$f_C = \begin{cases} \gamma' + \kappa' \log_{10}\left[\lambda'(\rho - \rho_{Med}) + 1\right], & \rho_{Min} < \rho < \rho_{Med} \\ \gamma'' + \kappa'' \log_{10}\left[\lambda''(\rho - \rho_{Med}) + 1\right], & \rho_{Med} < \rho < \rho_{Max} \end{cases}, \tag{3.11}$$

where λ' and λ'' drive the exact shape of the cost functions. These allow the operator to either force the solution to stay close to the control curve (parameters set to high values) or tolerate solution far from this (parameters set to low values). In Fig. III.5, the continuous line corresponds to $\lambda' = \lambda'' = 200$, which represents the default setting. The other four parameters in (3.11) are set by enforcing the constraints on the borders:

$$\begin{cases} \gamma' = 0.5 \\ \gamma' + \kappa' \log_{10}\left[\lambda'(\rho_{Min} - \rho_{Med}) + 1\right] = 0 \end{cases} \quad \begin{cases} \gamma'' = 0.5 \\ \gamma'' + \kappa'' \log_{10}\left[\lambda''(\rho_{Max} - \rho_{Med}) + 1\right] = 0 \end{cases}. \tag{3.12}$$

Expression (3.11) is used to evaluate the cost function for a combination *D-B* of delay and capacity usage. After computing the angle ϑ by means of (3.10), ρ_{Med} comes out from the conversion of (3.9) in the polar plane:

$$\rho_{Med} = \frac{\sin\vartheta - \beta\cos\vartheta \pm \sqrt{(\beta\cos\vartheta - \sin\vartheta)^2 - 4\alpha\chi\cos^2\vartheta}}{2\alpha\cos^2\vartheta}. \tag{3.13}$$

If ρ is higher than this value, ρ_{Max} has to be computed as follows:

$$\rho_{Max} = \begin{cases} \sqrt{B^2 + J^2}, & 0 \le \vartheta \le \pi/4 \\ \sqrt{D^2 + I^2}, & \pi/4 \le \vartheta \le \pi/2 \end{cases}. \tag{3.14}$$

Differently, if $\rho < \rho_{Med}$, ρ_{Min} has not to be computed being always equal to *Q*.

It is important to underline that the expression of (3.11) is not dependant on how the two regions and the curve are described, but only on the value of ρ in the area borders and the curve.

Genetic Operators

Always referring to Fig. I.10, the solution is found by making the population evolving by means of crossover and mutation operators. In particular, the most fitting part of the population P_{best} was selected and directly inserted in the new generation, while the rest of the population P_{worst} was discarded and replaced by a sub-population created by means of the crossover and mutation operator. A crossover operator (with a probability $CROSS$) was used to interchange the elements of two strings, while mutation operator (with a probability MUT) tried to lead the search out of local optima. In the case of two identical chromosomes resulted after the crossover and mutation operations, two individuals are randomly generated.

Parameters

With respect to Fig. I.10, the termination condition was satisfied once *g* reached a selected number of iterations (IT) or the fitness function maintained the same value for IT_{MAX} iterations.

3.3. Complexity Analysis

The resulting time complexity does not depend on the used cost function and is expressed by: $O(Dijk + Path + Tree + Gen)$, where:

$$\begin{cases} O(Dijk) = O(N^3) \\ O(Path) = O(S \cdot K_z{}^2 \cdot N) \\ O(Tree) = O\left(S \cdot (H_N \cdot \ln N)^{K_z}\right) \\ O(Gen) = O(IT \cdot S \cdot K_z \cdot P_{\dim} \cdot N^2) \end{cases} . \quad (3.15)$$

In particular, $O(Dijk)$ is the complexity of the Dijkstra algorithm, $O(N^2)$, applied to all N nodes. $O(Path)$ represents the complexity for generating additional paths: in fact, all K_z destination nodes need to be analyzed for all S sessions and new paths are searched starting from the best ones previously found. $O(Tree)$ is the complexity for the tree generation procedure: the maximum number of paths found for each destination node is $H_N \cdot \ln N$, i.e. the upper bound for the number of $\mathbf{L}_z$ arrays; this procedure is iterated for all K_z nodes and has to be repeated for all S sessions. $O(Gen)$ represents the genetic algorithm complexity, considering $N \times N$ matrices and K_z destination nodes for all S sessions.

Even though the complexity quickly increases as network and destination nodes number grows, it has to be highlighted that these are intended as routers, and not as end-stations. For this reason, high N values (i.e. more than 50) and K_z values (i.e. more than 20) are very uncommon to be reached.

3.4. Experiments

Settings

For presentation convenience let's refer to the GA algorithm using cost functions f_B and f_C with GA-B and GA-C, respectively.

In this Section, the results for different network configurations are presented. The network topologies used in the simulations were generated by a random graph model proposed by Waxman [W88]. The generator first randomly distributes N nodes over a square coordinate grid. The link between any two nodes n_i and n_j is added by the probability function $P((n_i, n_j)) = \beta_w \exp(-d_w(n_i, n_j)/\alpha_w \cdot l_w)$, where $d_w(n_i, n_j)$ is the Cartesian distance between nodes n_j and n_j, l_w is the maximum possible distance between any two nodes and parameters α_w and β_w are real numbers in the range (0,1]. Note that these parameters can be appropriately tuned to obtain the desired characteristics in the resulting graph. In the simulations, α_w = 0.4 and β_w = 0.5

are set for N=20 nodes. The graphs are used only when connected: if generated graph are not connected, it is discarded.

The results were compared with the solutions calculated using the cost functions proposed by Wang [WLJ02] and Chen [CGY00]. It has to be stressed that they were calculated by using genetic algorithms and not by their heuristics: thus referring to them as GA-Wang and GA-Chen, respectively. This choice is made since the link cost adopted by Chen and Wang have a different meaning respect to the cost function proposed in this Chapter thus the optimal value and the time processing have no sense to be compared, while it is quite important to analyze the traffic distribution (in terms of bandwidth and delay) in the network. As proposed by Wang, the link cost adopted for these comparisons was assigned to be the distance between two end nodes of that link (and normalized when running Chen algorithm), while the capacity constraint for each link was assigned to be the link cost times a random number in [2/N, 1].

The member node sets are all equal (i.e., $\mathbf{M}_1 = \mathbf{M}_2 = \ldots = \mathbf{M}_z = \mathbf{M}$) and the number of multicast sessions is equal to the size of member node set (i.e., $z = |\mathbf{M}|$), always for creating test conditions similar to Wang ones.

The control curve has been set with the following D-B control points: (0.8, 0), (0.5, 0.4) and (0, 0.8) obtaining, from (3.9), the values of the parameters(α, β, χ) = (-0.4667, -0.6670, 0.8) drawing the curve depicted in Fig. III.6; the good and bad areas borders are set with $I = J = 0.9$ and $Q = 0.2$; the cost function behaviour is the default one with $\lambda' = \lambda'' = 200$.

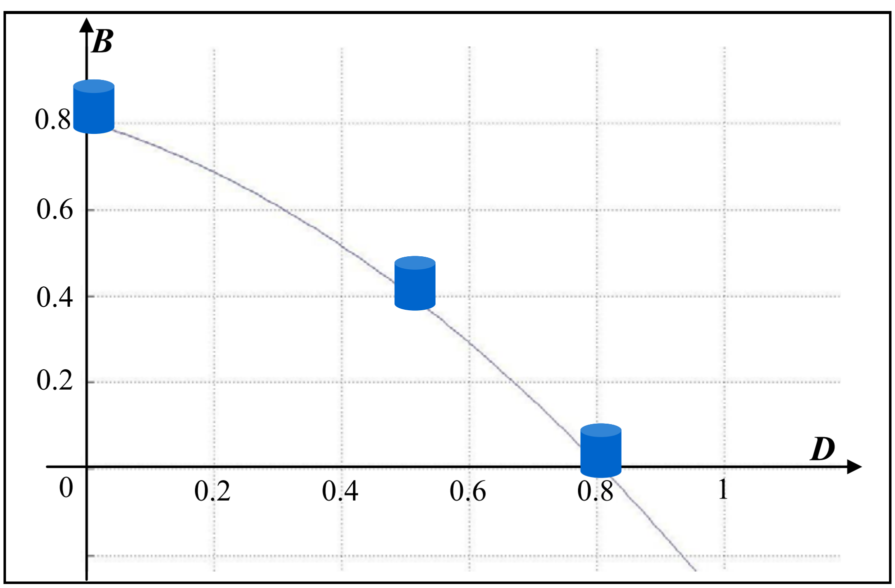

Fig. III.6. Control curve for the adopted control points

Test analysis

First of all, some tests were run for analyzing the dependence of GA-parameters respect to the results obtained: it was experimented P_{Dim} varying in the range (8, 16, 32, 64), while $CROSS$ and MUT in the range 0.3÷0.7 and 0.01÷0.3 respectively. The maximum gap among all the solutions was observed to be less than 2%. Then, if such this error can be tolerated, every choice bring to a good solution, otherwise the following consideration have to be taken into account. An high population size brings to better solutions at the expense of a higher processing time; the MUT parameter is suggested to be set equal or higher than 0.1 for avoiding an excessive number of iterations, while the $CROSS$ parameter does not sort significant effects in the range used. For these reasons, in the following experiments the adopted parameter values are: $CROSS$ = 50%, MUT = 30%, $P_{Dim} = 32$, $IT = 1000$, and $IT_{MAX} = 200$.

To observe the behaviour of the proposed algorithms in critical conditions, the network was loaded with background traffic near 70%. In particular, the background unicast traffic occupation for each link was randomly set in the range 50% - 80%, while the link propagation delay π_{ij} was randomly set in the range 1 - 40 msec in each link. When applying GA-C, all flows have been treated evenly ($\alpha_z = 1$ and $y_z = 1$ for every z).

The results shown from Fig. III.7 to Fig. III.10 are referred considering z and K_z in the range 2÷7 (where $z = K_z = M$ as stated in the previous subsection).

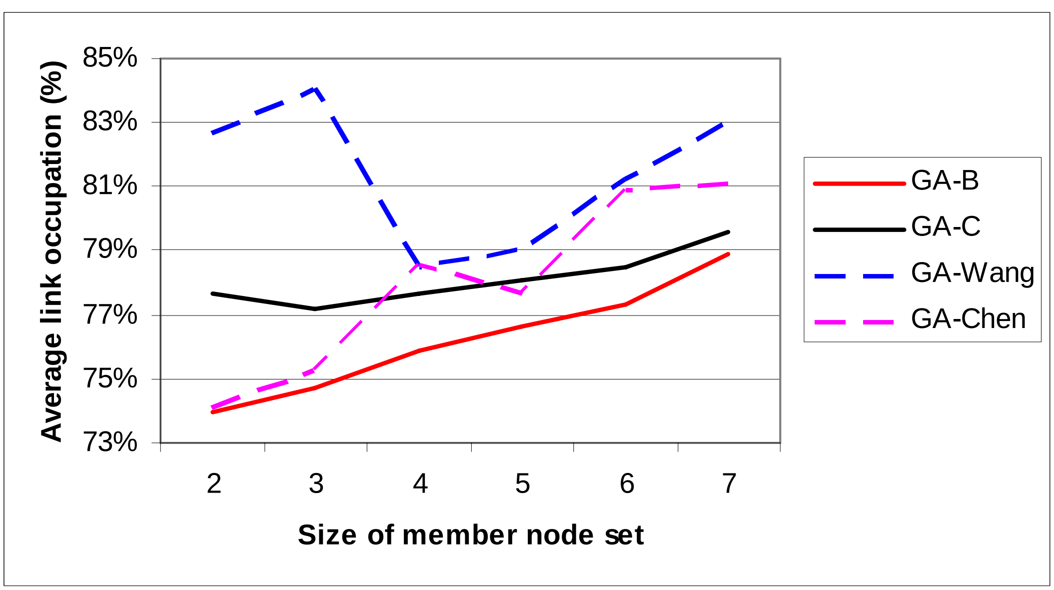

Fig. III.7. Multicast links occupation

As shown in Fig. III.7, the GA-B algorithm (straight red line) obtained the best results in terms of bandwidth occupation in the links used by the multicast sessions to the detriment of some losses in the end-to-end delays values (Fig. III.8) where GA-B have low values, but not the best ones. The GA-C algorithm (straight black line) used less effectively the bandwidth, but it showed absolutely the best results in the end-to-end delay analysis.

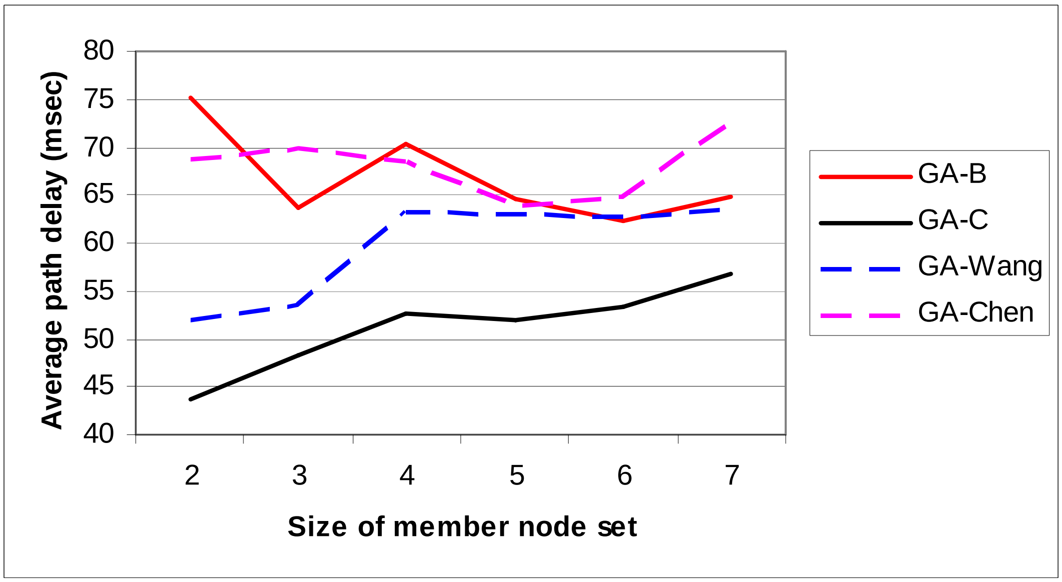

Fig. III.8. End-to-end delay

Always referring to Fig. III.7 and Fig. III.8, the Wang algorithm (and then GA-Wang, the sketched blue line) had its scope on using the links with the lowest weight (and not necessary the bandwidth) and was not interested in optimizing this usage (as highlighted also by the first values in Fig. III.7) even though the end-to-end delay was, for few sessions, lower than GA-B one (Fig. III.8); the Chen algorithm (and then GA-Chen, the sketched fuchsia line) was interested on reducing the traffic on the most loaded link thus leading to a better usage of the link occupancy (even not so better than the GA-B) but not of the delays requirements where GA-Chen showed the worst results.

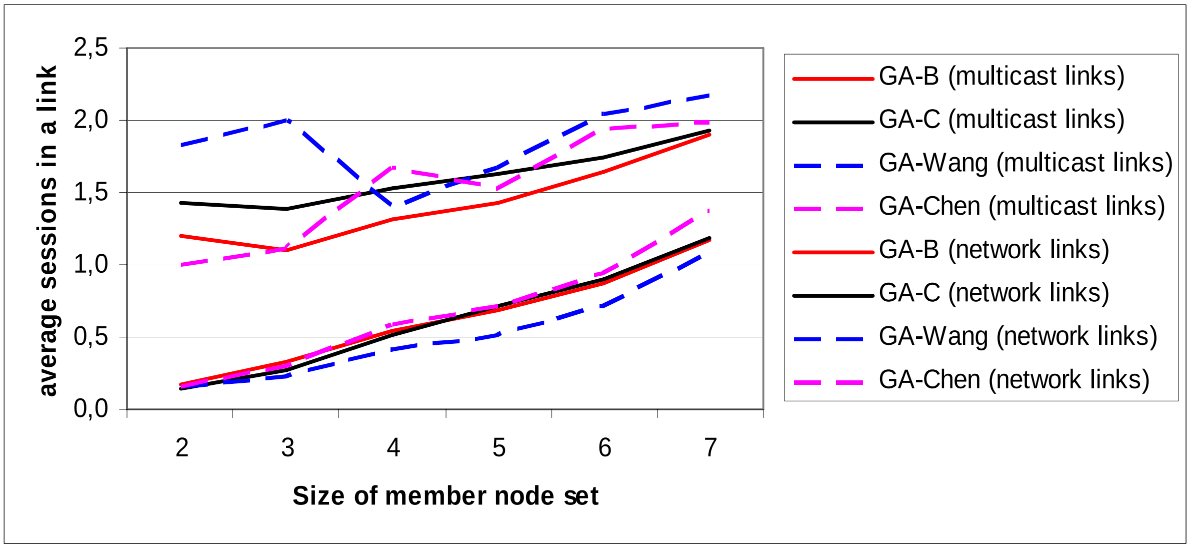

Fig. III.9. Number of sessions in the links (network and multicast)

The optimal usage of the multicast links led to a tree growth (in terms of spanning or hop increase) as shown in Fig. III.9 (upper set of curves): the sessions over the links used by GA-B and GA-C were less than GA-Wang (which tend to pack the traffic flows) and GA-Chen ones, but greater when considering the sessions over all the network links (lower set of curves in Fig. III.9) because GA-B and GA-C used more links (more spanning or more hops), and the total traffic generated was greater too.

However, this traffic growth is not so relevant: as the Fig. III.10 shows, the proposed algorithms create less than 1% traffic more than GA-Wang algorithm, while have similar or lower results respect to the GA-Chen algorithm. Referring to the proposed algorithms, this spanning phenomenon is obviously lower in GA-C algorithm, since too many hops lead to higher delays.

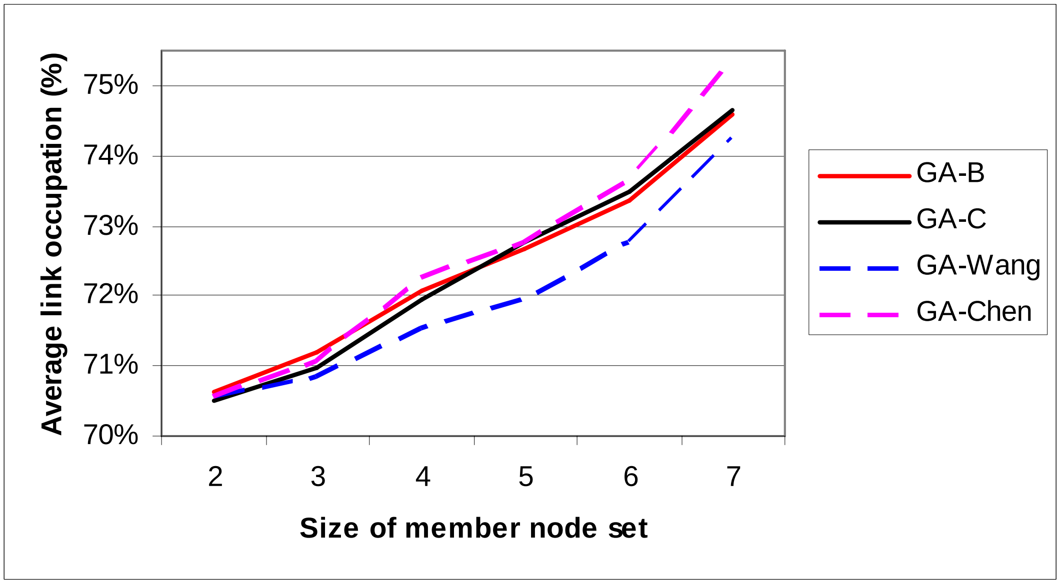

Fig. III.10. Network links occupation

In order to show that a good solution can be obtained even not using DFS during the paths search (when building the space solution), in Table III.1 were shown the gap ratios of the objective value of GA-B and GA-C to the objective value GA-DFS found by the DFS algorithm (applied instead of Dijkstra and additional path computation): the gap ratio for a certain parameter is defined to be the differences between the value obtained with the proposed method and the one obtained by using GA-DFS, all divided for the GA-DFS value (and multiplied for one hundred); the measured parameters were the end-to-end delay, multicast link occupation, network link occupation, cost and processing time.

Table III.1. Comparison of the ratio to DFS for the proposed genetic algorithms GA-B and GA-C

Number of sources	End-To-End delay		Multicast link occupation		Network link occupation		Cost		Processing time	
	GA-B	*GA-C*	*GA-B*	*GA-C*	*GA-B*	*GA-C*	*GA-B*	*GA-C*	*GA-B*	*GA-C*
2	0,00%	-6,91%	0,00%	3,50%	0,00%	-0,08%	0,00%	0,40%	-8,55%	-15,02%
3	-18,47%	-2,90%	0,43%	1,23%	0,00%	-0,16%	0,46%	0,05%	-31,52%	-14,75%
4	-10,08%	1,57%	0,44%	0,71%	0,00%	0,16%	0,47%	0,22%	-46,68%	-41,68%
5	-20,55%	2,13%	0,39%	0,00%	-0,16%	0,08%	0,10%	0,20%	-75,06%	-80,86%

It has to be noted that costs were quite similar (differences under the 0.5%), in spite of relevant processing time growth. The other results (delay and link occupation) showed also that differences above 4% are always negative, i.e. occurred when the proposed algorithms performed better (even not in terms of cost). It is important to underline that the results for more than five sources were not shown since memory allocation was too high. The goodness of the solution due to the usage of GA technique is not proved here because the historical use of this approach and its proven goodness.

Finally, in Fig. III.11 and Fig. III.12, it was analyzed the possibility, provided by the GA-C algorithm, to give more priority (in terms of end-to-end delay requirements) to a traffic flow than the others. Leaving the same settings adopted up till now, only fixing z and K_z to the maximum value (i.e. there were seven sources with seven destination nodes each) and let α_z varied from 1 to 100 for the first (Fig. III.11) and the fourth (Fig. III.12) sessions only.

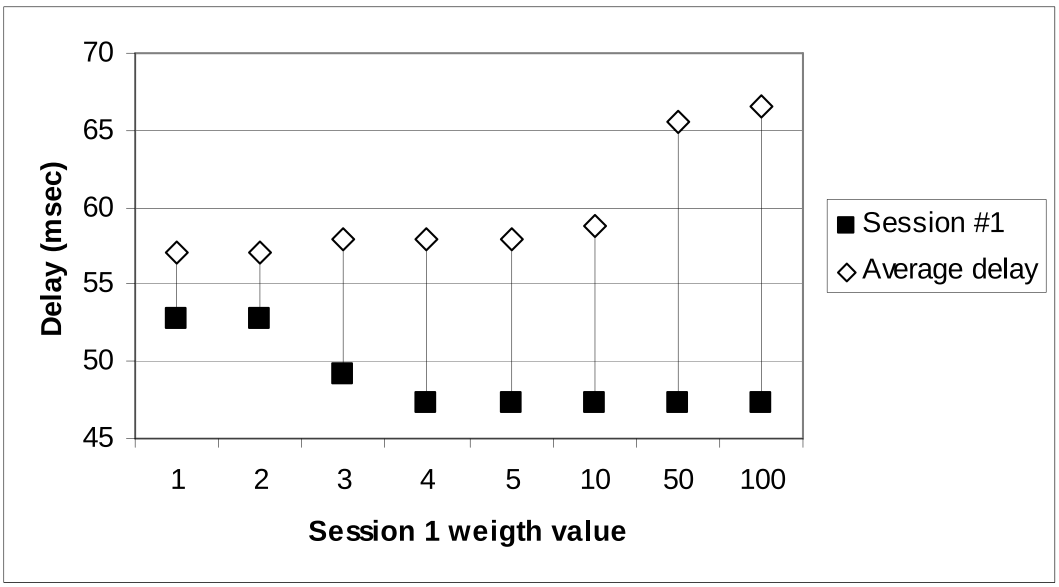

Fig. III.11. Traffic priority for session 1

In both figures these sessions reached their minimum end-to-end delay value since for low weight values both in cases their delay value was initially under (Fig. III.11) or above (Fig. III.12) the average. It is interesting to observe that an excessive increase does not provide further improvement while having the backside effect to increase the delay of all the others sessions (and thus leading to an average and undesired delay increase).

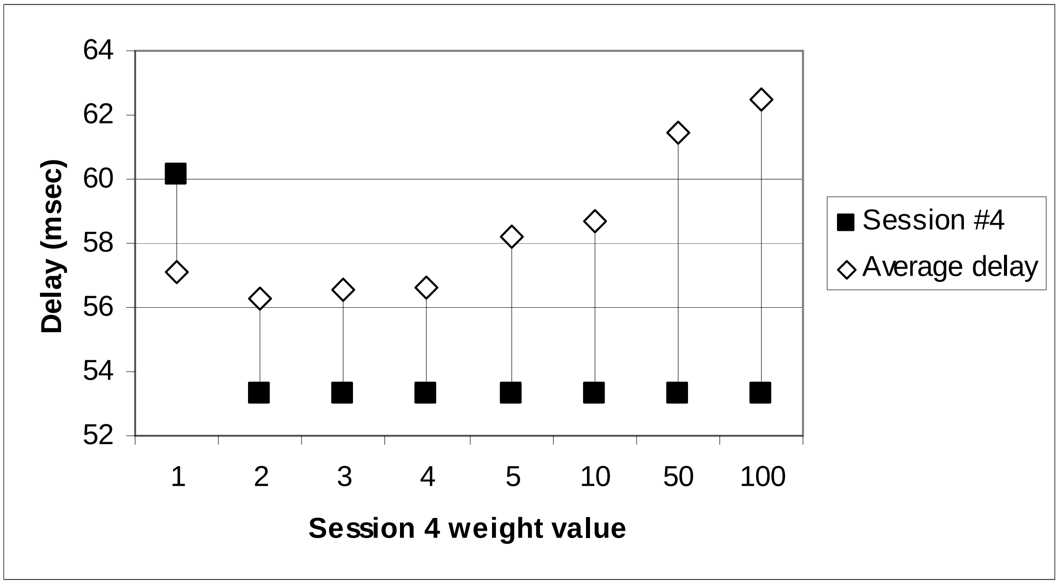

Fig. III.12. Traffic priority for session 4

Chapter IV

Power optimization for Wi-Fi stations

The IEEE 802.11 [Wi99] standard (see Chapter I, Section 1.2) specifies a Power Management (PM) mechanism (see Chapter II, Section 2.2)that allows a mobile station to enter in a state of low power consumption (doze) when its interface is idle. Much research has been conducted on PM and some inefficiencies and limitations of this mechanism have been revealed. Several solutions have been proposed to overcome such problems. Most of these refer to the IBSS mode; among these, in [BCD01] the authors propose a Distributed Contention Control mechanism to guarantee the optimal power consumption, whereas a dynamic choice of the ATIM (Announcement Traffic Indication Map) window size in [JV02]. In [THH02] authors have addressed the power management in MANET (Multi Hop Ad Hoc Networks). On the contrary, only few works consider the infrastructure Basic Service Set (iBSS) system. In [CSE04] and [T00] authors propose application dependent solutions to improve the efficiency in power management. The authors in [LN05] introduce a scheme in which the AP schedules the transmission sequence of buffered packets by piggybacking a bit map in TIM (Traffic Indication Map). In [CH04] the listen interval is adjusted according to a power management policy that keeps into account frame loss and delay.

The basic idea is that the PM algorithm used in iBSS may still be improved with a shrewd management of the stations status. Furthermore, these improvements should go towards an application-independent approach so as not to increase the algorithm complexity. Accordingly, in this Chapter, is presented a new strategy that introduces significant changes in the AP behaviour, which acquires a high decisional power.

First of all, it was investigated the total average energy consumption necessary to successfully receive a frame for the stations that, being in doze state, are woken up by the AP. This analysis puts in evidence that such energy strongly depends on the number of stations contending the channel: this means that in case of high traffic, the stations to be woken up could incur into long waiting times that can negatively affect the performance of the system, introducing delays and leading to waste of power. For this reason, a simple but effective PM function is proposed aiming at minimizing these waiting times to reduce the power consumption of the overall system. It is based on giving the AP the authority to decide whether a station in doze mode with pending frames should be woken up or not, weighting the energy necessary to receive a frame and the latency of frames to be sent. In this way, the AP may defer the waking up of some stations to a subsequent interval so as to reduce the time spent in active mode for the woken stations and, as a direct consequence, the power waste.

The next rest of the Chapter is organized as follows. Section 4.1 presents the proposed approach to improve the PM efficiency. Section 4.2 is used for providing numerical results and conclusions.

4.1. Proposed approach

The proposed Power Management strategy acts as follows. At each beacon interval, the AP monitors the queues of the doze stations and, on the basis of a particular cost function, decides which stations with pending frames have to be woken up, putting their correspondent ID in the TIM. This means that, differently from the IEEE 802.11 protocol, some stations with pending frames could be not woken up, deferring the transmission of such frames to successive beacon intervals. Obviously, the cost function has to take into account the introduced latency: if the AP decides to keep in doze state a station with pending frames, this will suffer an additional latency equal to the length of a beacon interval. In an application-dependent environment, this latency plays a fundamental role that has to be considered ([CSE04], [T00], [WFB04]).

Next subsection provides the description of the system and the notation used; the second subsection provides the analysis of the average energy consumed by a woken station for the receipt of a frame as a function of the number of contending stations. The cost function is provided in the last subsection.

System description

This subsection describes a time-discrete model that will be adopted to evaluate the performance of the proposed strategy and to define the cost function. Consider a wireless cell controlled by an AP with M stations implementing the PM. Let's index i and k the stations and the beacon intervals, respectively. At the beginning of each beacon interval k, the AP lists in the TIM the ID's of the stations to be woken up. Let vector $\mathbf{\Lambda}_k = \{\lambda_k^i, i = 1,...,M\}$ represent the PM decision of the AP at the beginning of interval k:

$$\lambda_k^i = \begin{cases} 1 \text{ if the AP wakes up station } i \\ 0 \text{ otherwise} \end{cases}.$$

Accordingly, the number of stations woken up is $n_k = \|\mathbf{\Lambda}_k\|_1$. Note that λ_k^i has to be zero for the active stations and for stations in doze state without pending frames. To take into account the state of stations at the beginning of interval k vector $\mathbf{B}_k = \{\beta_k^i, i = 1,..., M\}$ has been introduced:

$$\beta_k^i = \begin{cases} 1 \text{ if station } i \text{ is in doze with pending frames} \\ 0 \text{ otherwise} \end{cases}.$$

The number of stations that the AP wakes up in the IEEE 802.11 PM is $m_k = \|\mathbf{B}_k\|_1$. In fact, the standard states that all the stations in doze with pending frames have to be woken up.

Additionally, let $\mathbf{A}_k = \{\alpha_k^i, i = 1,..., M\}$ represent the evolution of the state during a beacon interval:

$$\alpha_k^i = \begin{cases} 1 \text{ if the AP receives frames for station } i \text{ in doze} \\ -1 \text{ if station } i \text{ wakes up by itself} \\ 0 \text{ if station } i \text{ maintains its state} \end{cases}.$$

In order to take into account the latency introduced by the proposed algorithm let also introduce vector $\mathbf{D}_k = \{d_k^i, i = 1,..., M\}$, which represents the number of beacon intervals during which the transmission of pending frames have been delayed.

Based on the previous definitions, the system evolves according to the following expressions:

$$\mathbf{B}_{k+1} = \mathbf{B}_k + \mathbf{A}_k - \mathbf{\Lambda}_k, \tag{4.1}$$

$$d_{k+1}^i = \begin{cases} d_k^i + \beta_k^i & \text{if } \lambda_k^i = 0 \\ 0 & \text{if } \lambda_k^i = 1 \mid \alpha_k^i = -1 \end{cases}, \tag{4.2}$$

with the following constraints:

$$\begin{cases} \mathbf{\Lambda}_k \le \mathbf{B}_k \\ \vec{\mathbf{0}} \le \mathbf{A}_k + \mathbf{B}_k \le \vec{\mathbf{1}} \\ \mathbf{D}_k \le \vec{\mathbf{1}} \cdot d_{\max} \end{cases}, \tag{4.3}$$

where $d_{\max}$ is an integer parameter to be chosen on the basis of the maximum tolerable latency (depending on the application) for frames and the AP buffers size. Finally, μ_k denotes the number of active stations in the cell with frames to transmit.

Modelling

The scope of this subsection is to determine the total average energy necessary for the successful receipt of a frame for a station being woken up by the AP; this energy will be adopted to define the cost function that drives the proposed power management strategy. In the following, the formulas necessary to draw this energy are introduced: the same modelling shown in [WA04] was substantially adopted with some key differences that will be underlined.

Time is discretized by slot of length T_{slot}. When a station starts the collision avoidance mechanism, randomly selects the backoff counter (hereafter BC) included in the range $[0, CW(j)]$ where $CW(j)$ is the contention window after j unsuccessful transmission attempts. For the first attempt the contention window assumes the minimum value $CW_{\min}$; this value is doubled for each collision till the maximum value $CW_{\max} = 2^w CW_{\min}$ is reached. In this research, the hidden station phenomena is neglected assuming that all stations can hear transmissions of each other

Consider a wireless cell with n contending stations with at least one frame to transmit/receive. Differently from [WA04], the reference of the analysis is focused on a doze station that, being woken up by the AP, has to contend the channel access with the other n-1 stations (among which some are active and some others that have been woken up) to receive the frames buffered at the AP. The total average energy is given by the sum of four main factors: the energy used during the backoff stages E_{BC}, the energy waste due to collisions E_C, the energy spent overhearing the other transmissions E_{fr}, and the energy necessary to successful receive a frame from the AP E_{tx}. To define these terms, some useful expressions need to be introduced. The probability that a transmission incurs collision p can be approximated as in [TC01] with $p = 1 - (1 - 1/E[BC])^{n-1}$, where $E[BC]$ is the expected value of the backoff counter:

$$E[BC] = 0.5 \cdot CW_{\min}\left(1 - p - p(2p)^w\right) \cdot (1 - 2p)^{-1}. \tag{4.4}$$

Furthermore, the probability that the reference station suffers a certain number N_C of collisions before successfully transmitting can easily be written as:

$$\Pr\{N_C = i\} = p^i(1 - p). \tag{4.5}$$

The expectation of such a probability is thus given by:

$$E[N_C] = \sum_{i=0}^{\infty} i \cdot p^i (1 - p) = \frac{p}{1 - p}. \tag{4.6}$$

When a collision occurs, the contending stations will start decrementing their BCs again after a DIFS interval, which is the time T_C during which the channel remains busy for a collision. It is now possible to define the first two contributes to the total average energy:

$$E_{BC} = P_{idle} \cdot (E[N_C] + 1) \cdot E[BC] \cdot T_{slot}, \tag{4.7}$$

$$E_C = P_{idle} \cdot E[N_C] \cdot T_C, \tag{4.8}$$

where P_{idle} is the power consumed by the station during the idle state. To derive E_{fr}, the average number of overheard transmissions by the tagged station during its backoff stages need to be defined:

$$\overline{N}_t = (E[N_C]+1) \cdot E[BC] \cdot P_{tr}, \tag{4.9}$$

where P_{tr} is the probability that any number of the n-1 occupies the channel for a transmission and is equal to p. Among these overheard transmissions, $\overline{N}_t \cdot P_s$ will be successful and $\overline{N}_t \cdot (1-P_s)$ will be unsuccessful due to collisions, where P_s is the probability that any of these transmissions is successful:

$$P_s = P_{tr}^{-1} \cdot (n-1) \cdot E[BC]^{-1} \left(1 - E[BC]^{-1}\right)^{n-2}. \tag{4.10}$$

The energy spent in overhearing the other transmissions is thus given by:

$$E_{fr} = \overline{N}_t [P_s T_s + (1-P_s) T_C] \cdot P_{idle}, \tag{4.11}$$

where

$$T_s = \begin{cases} DIFS + T_{frame} + SIFS + T_{ACK} \\ \text{for an active station} \\ DIFS + T_{POLL} + 2SIFS + T_{frame} + T_{ACK} \\ \text{for a woken station} \end{cases} \tag{4.12}$$

is the duration of a successful transmission and T_{POLL}, T_{frame}, T_{ACK} are the transmission times for the PS-Poll frame, the data frame and the ACK frame, respectively. Differently from [WA04], this dissertation takes into account also the duration of a successful transmission for a woken station that has to retrieve a buffered frame from the AP. The energy consumption for a successful transmission is then different from that adopted in [WA04]:

$$E_{tx} = P_{tx} \cdot T_{POLL} + P_{idle} \cdot (DIFS + 2SIFS) + P_{rx} \cdot (T_{frame} + T_{ACK}), \tag{4.13}$$

where P_{tx} and P_{rx} represent the power consumed by the woken station in transmission and receipt states, respectively.

Then, the total average energy necessary for the reference station to successfully receive a frame from the AP is provided by:

$$\overline{E} = E_{BC} + E_C + E_{fr} + E_{tx}. \tag{4.14}$$

This term strongly depends on the number of contending stations and, in an environment with high power constraints, has to be as low as possible. Fig. IV.1 draws $\overline{E}$ as a function of the contending stations, whit system parameters set according to Table IV. It is worth noting that this curve has a linear behaviour. Applying a least-squares regression to $\overline{E}$, a straight line with slope

$a = 0.0048$ and offset $b = 0.0028$ was obtained, with very low norm of the residuals equal to 0.0011.

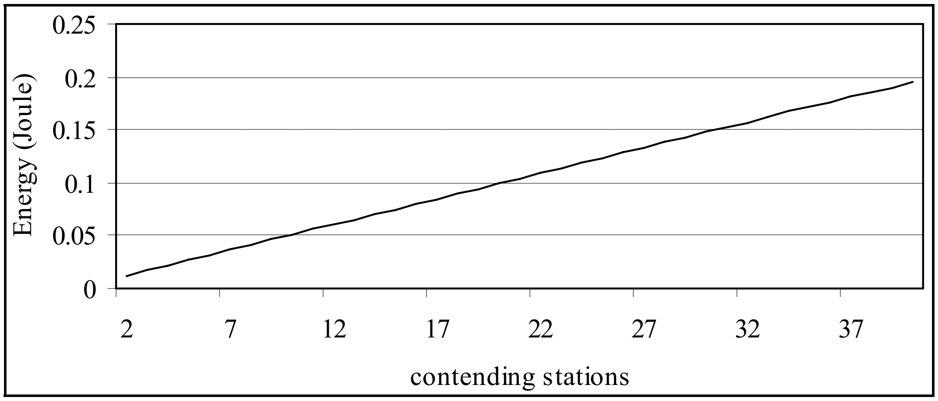

Fig. IV.1. Total average energy $\overline{E}$, computed with settings parameters of Table IV.

Cost function

The objective of proposed PM algorithm is to find out a good compromise between energy consumption and frame latency. To this, the cost function is defined as follows:

$$F = W_1 + \vartheta \cdot W_2, \tag{4.15}$$

where W_1 and W_2 measure energy consumption and frame latency, respectively. The parameter $\vartheta > 0$ is a weight parameter whose importance will be discussed in the following. At the beginning of every interval *k*, the AP seeks for the solution $\mathbf{\Lambda}_k$ that minimizes (4.15) subject to the constraints in (4.3). The term W_1 measures the energy consumption of the woken stations at the beacon *k* according to equation (4.14). Clearly, this term weights the choice of the AP to wake up a certain number n_k of stations in doze. Accordingly, the used expression follows:

$$W_1 = n_k \cdot \overline{E}(n_k + \mu_k). \tag{4.16}$$

As to W_2, it measures the latency of pending frames addressed to stations that have not been woken up by the AP:

$$W_2 = T_b \cdot \|\mathbf{D}_{k+1}\|_1. \tag{4.17}$$

To make a comparison between the proposed strategy and the standard one, the expression used to evaluate the performance of the PM implemented in the IEEE 802.11 protocol is:

$$F_k^{PM} = m_k \cdot \overline{E}(m_k + \mu_k), \tag{4.18}$$

given that there are not deferred transmissions in the standard, the term W_2 is always equal to 0.

The parameter ϑ controls the optimal balance between energy consumption and latency and its setting is a tricky task. When set to high values, the proposed algorithm converges toward the standard strategy, where the doze stations with pending frames are always awaken. Contrarily, low values bring to frequent postponements of frame transmissions with the aim of reducing station energy consumption. In order to understand the importance of the energy term with respect to the delay one, it should be considered one beacon interval latency per frame in terms of the energy dissipated by an active station during a beacon interval. The parameter ϑ should be set to $\overline{E}$ in case of equivalent importance. In what follows, the effects of ϑ selection on the proposed strategy are investigated and some general rules to guide the setting of the parameter are proposed. To this purpose, consider a linear approximation for $\overline{E}$ and assume that for each deferred transmission $d_{k+1}^{i} = D < d_{\max}$. The cost function for the standard and proposed strategies can be approximated as follows:

$$F^{PM} \cong m[a(m+\mu)+b], \tag{4.19}$$

$$F \cong (m-x)\cdot[a(m-x+\mu)+b]+\vartheta(T_b D)x \tag{4.20}$$

where $0 \le x \le m$ represents the number of deferred stations and the pair *a* and *b* come from the linear approximation. The difference between the two cost functions can be written as follows:

$$\Delta = ax^2 + (\vartheta T_b D - 2am - a\mu - b)x\,. \tag{4.21}$$

Let $x_{\min}$ denote the abscissa corresponding to the vertex of the parabola. This is the number of transmissions to stations in doze that the AP decides to postpone; the range of values of interest is 0÷*m*. As already noted, the greater ϑ, the lower $x_{\min}$ is. When the vertex is located at the origin, the AP wakes up all the doze stations with pending frames as in the standard strategy. This condition is reached when the weight parameter has been set to the maximum value:

$$\vartheta_{\max} = (T_b D)^{-1}\cdot(2am + a\mu + b). \tag{4.22}$$

By reducing ϑ the vertex of the parabola moves toward higher values of *x*; this means that there will be a threshold value ϑ_{th} such that for each value lower than this, the minimum value of Δ will be always $x_{\min} = m$, i.e., the AP defers all the doze stations with pending frames. This threshold can be obtained by imposing that the vertex is located at abscissa equal to *m*:

$$\vartheta_{th} = (T_b D)^{-1}\cdot(a\mu + b). \tag{4.23}$$

Then, the range for the control parameter and its connection with $x_{\min}$:

$$x_{\min}(\vartheta) = \begin{cases} m & \text{for } 0 \le \vartheta \le \vartheta_{th} \\ \dfrac{-\vartheta T_b D + 2am + a\mu + b}{2a} & \text{for } \vartheta_{th} < \vartheta \le \vartheta_{\max} \end{cases}. \tag{4.24}$$

This is the description the impact of the weight parameter on the proposed cost function; in this way the effects of a chosen ϑ are exactly known.

Another solution is to dynamically adapt ϑ to the network traffic conditions so as to make the AP behaviour evolving in accordance with the system. To this aim, the setting of the weight parameter is linked at a certain beacon interval *k* to the probability that in the next interval the number of active stations is higher than the current one. Indeed, when $\Phi = \Pr\{\mu_{k+1} \geq \mu_k\}$ is high, it is advantageous from an energy saving point of view to wake up the doze stations at the beacon *k*, so a high value of ϑ is chosen. Contrarily, when this probability is low, it is more helpful to defer the transmissions toward doze stations, which should encounter lower traffic in the future, thus reducing the overall energy consumption. A possible application of this criterion, is to force the deferring of the transmission only when $\Phi < \Phi_{th}$ with $x_{\min}$ decreasing with Φ. Now, consider typical system values: $\mu = 20$, $m = 5$, $T_b = 0.1\sec$, and $D = 2$; it brings to $\vartheta_{th} = 0.5$ and $\vartheta_{\max} = 0.7$. In this range, due to (4.24) a liner relationship between Φ and $x_{\min}$ is assured with a linear relationship between Φ and ϑ, that is what is proposed to use, as shown in Fig. IV.2. Note that when $\Phi > \Phi_{th}$ any value of the weight parameter greater than $\vartheta_{\max}$ can be used. In a real contest, the above probability is not known *a priori* and has to be estimated during the temporal evolution of the system on the basis of the observed number of the active stations at each past beacon interval. There are several algorithms aimed at solving this type of problems that can be indifferently adopted in this work.

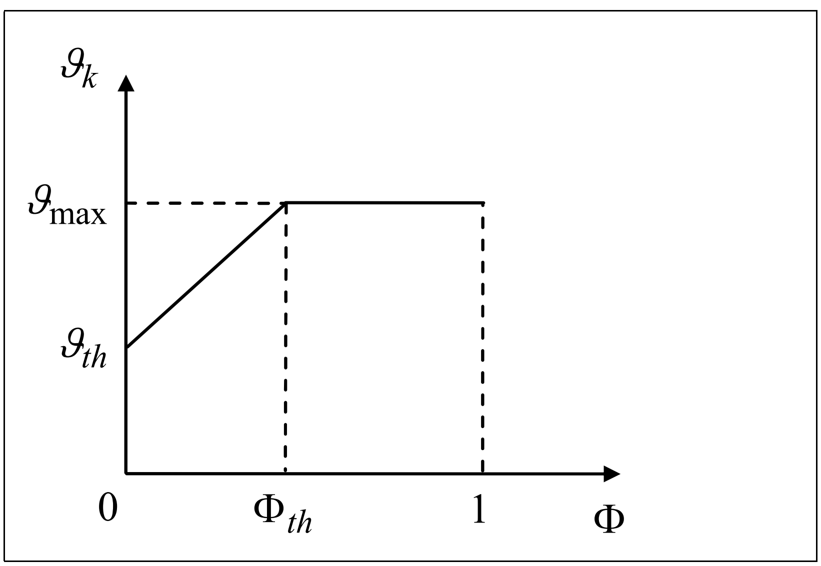

Fig. IV.2. Imposed relationship between ϑ_k and Φ

4.2. Numerical results and conclusions

To evaluate the performance of the proposed strategy, extensive simulations with Matlab were carried out. This Section refers to the results obtained with a wireless cell of M=40 stations, observed during N=1000 beacon intervals. The aim was to make a comparison of the performance, during the same run, between the standard and the proposed PM. The main system parameters settings are listed in Table IV.1.

Table IV.1. System Paramenters

$T_b = 100$ msec	$T_{frame} = 4$ msec
Tx Rate = 2 Mbps	$SIFS = 10$ µsec
$T_{slot} = 20$ µsec	$T_{ACK} = 56$ µsec
$T_{POLL} = 80$ µsec	$CW_{\min} = 128$
$T_{frame} = 4$ msec	$CW_{\max} = 1024$
$DIFS = 50$ µsec	$P_{tx} = 1.65$ Watt
$P_{rx} = 1.4$ Watt	$P_{idle} = 1.15$ Watt

The arrival process of packets addressed to each station is modelled with a Poisson point process with mean arrival rate equal to 5 packets per second; the probability for a station to switch from a state to another during a beacon interval is of 0.1. As to $d_{\max}$, it has to be set according to the delay tolerance of specific applications; in the simulations a value of 5 beacon intervals was used, corresponding to 500 msec. The number of active stations has been modelled as a birth-death model, with birth and death processes represented with Poisson and Erlang distributions, respectively, [FZ02] with an average number of active stations equal to 20. The threshold Φ_{th} is equal to 0.6: this means that only when Φ is lower than 0.6, the AP defers stations in doze with $d_{k+1} < d_{\max}$. Whenever $\Phi > \Phi_{th}$, the AP sets $\vartheta_k = \vartheta_{\max}$. Furthermore, a steady state condition was considered, i.e. the AP has already computed Φ. The function Φ was derived by constructing the histogram related to the occurrences of the number of the active stations at each beacon interval on the basis of the previous 1000. The term T_s in (4.12) is computed weighting the number of active and doze stations to be woken up.

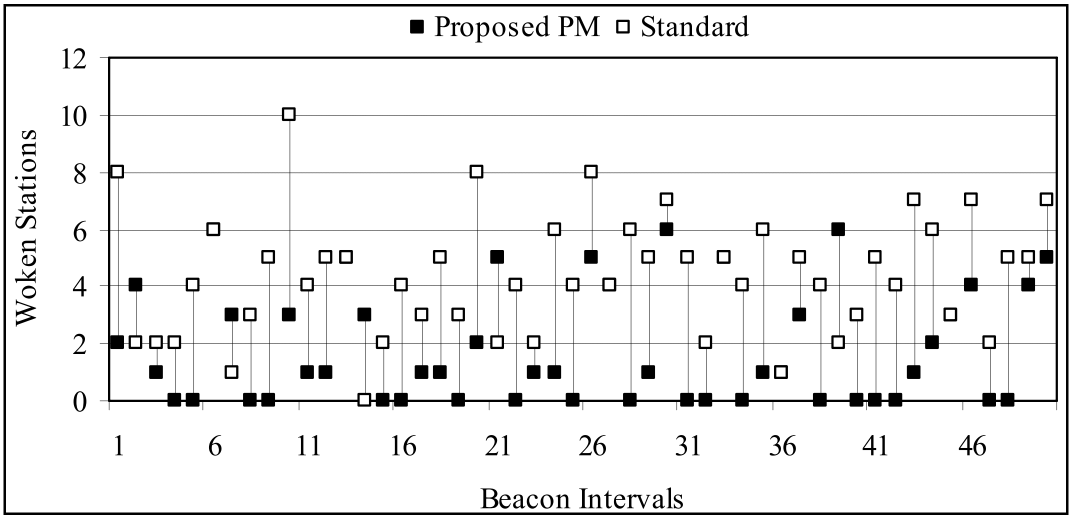

Fig. IV.3. Comparison of the number of woken stations between the standard and proposed PM

In Fig. IV.3 m_k and n_k are compared for the first 50 beacon intervals. As expected, the proposed strategy tends to wake up a number of stations lower than that triggered by the standard PM for almost all the beacon intervals. This fact can be easily explained by means of Fig. IV.4 where there is the comparison between the number of frames, directed to woken up stations, sent by the AP using the proposed strategy and the standard one.

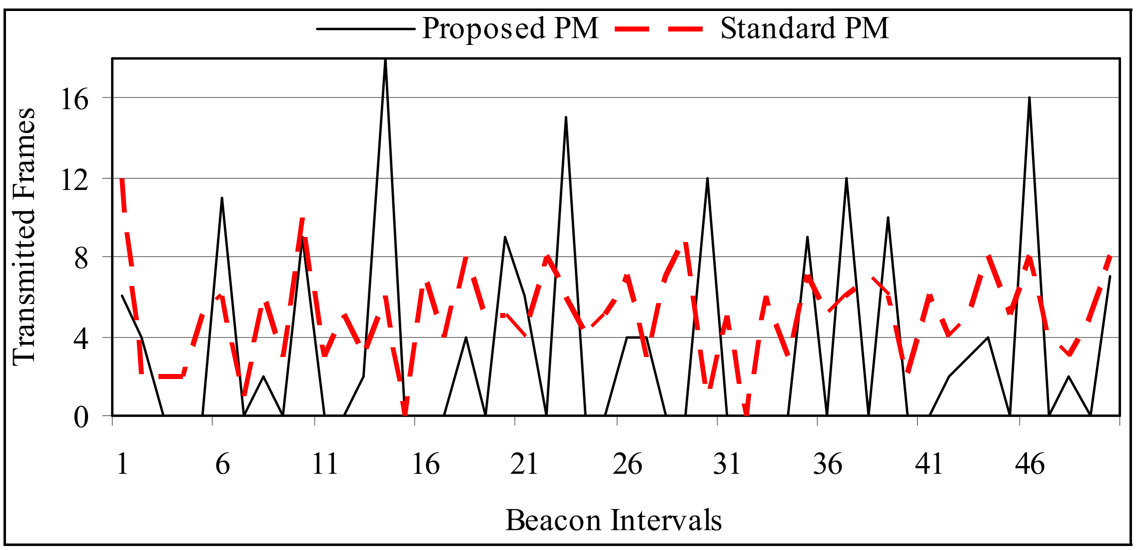

Fig. IV.4. Comparison of the transmitted frames from the AP between the standard and the proposed PM

As you can see in Fig. IV.4, in the proposed strategy, the AP aggregates a high number of frames to send at certain beacon intervals, this allows the AP to wake up a lower number of stations with more frames to receive. This fact leads to the generation of a more bursty traffic than the standard PM with the same average value equal to 4.5. Such trend is strictly linked to the behaviour of the active stations.

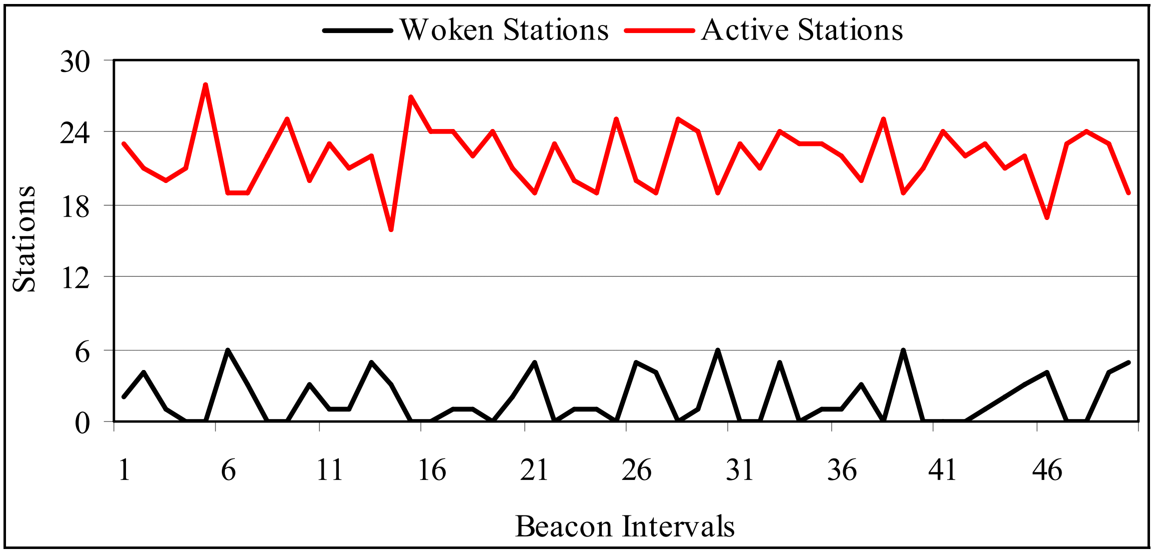

Fig. IV.5. Comparison between the woken and the active stations using the proposed PM

Fig. IV.5 shows the comparison between the number of active and woken up stations: due to the appropriate choice of the weight parameter among the beacons, in correspondence of the minimum values of the active stations there is the maximum values of the woken up stations and vice versa. This means that the proposed strategy adequately reacts to the temporal changes of the system and tends to wake up more stations when the network load is lower.

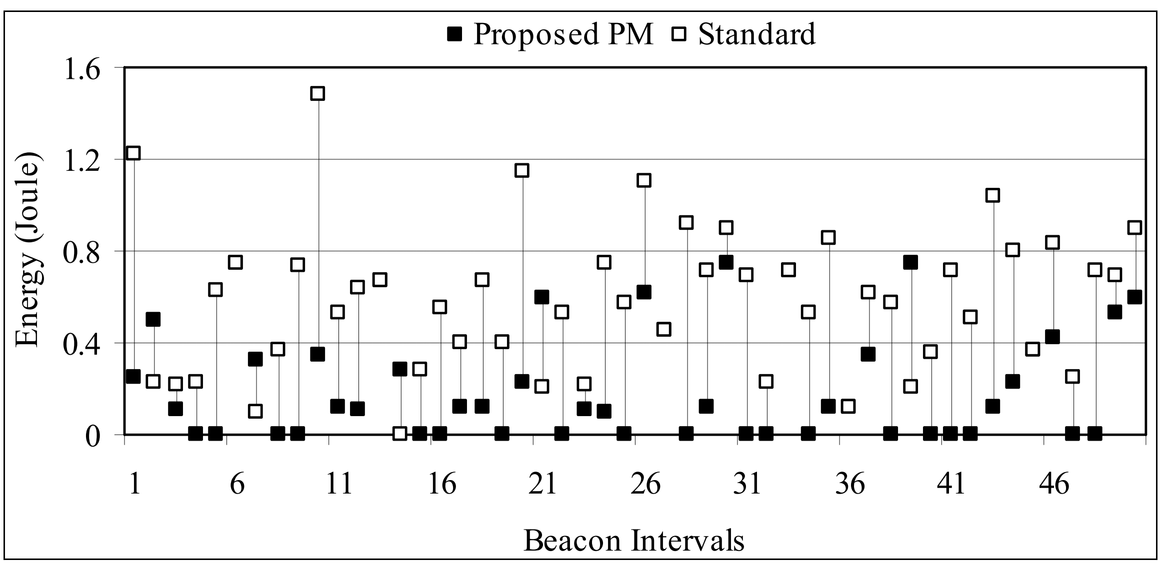

Fig. IV.6. Comparison of the total average energy consumption of the woken stations between the standard and proposed PM

Fig. IV.6 provides a comparison of the total average energy $\overline{E}$ of the woken stations between the proposed and the standard PM as computed by means of expressions (4.16) and (4.18), respectively. Note that there is a significant mismatch between the two functions for almost all beacon intervals; in fact, an average reduction of $\overline{E}$ of about 63% has been obtained. The drawback of this reduction is the increase in frame latency in the AP queues. The average latency obtained in this run is equal to 0.1472 sec.

A further counterpart is the higher buffer occupancy due to the deferred frames depicted in Fig. IV.7. Note that there are no deferred frame transmissions when using the standard PM. These results underline that, depending on the requirements of the wireless cell, the proposed strategy could achieve higher energy saving than the standard but introducing higher delays that can be controlled.

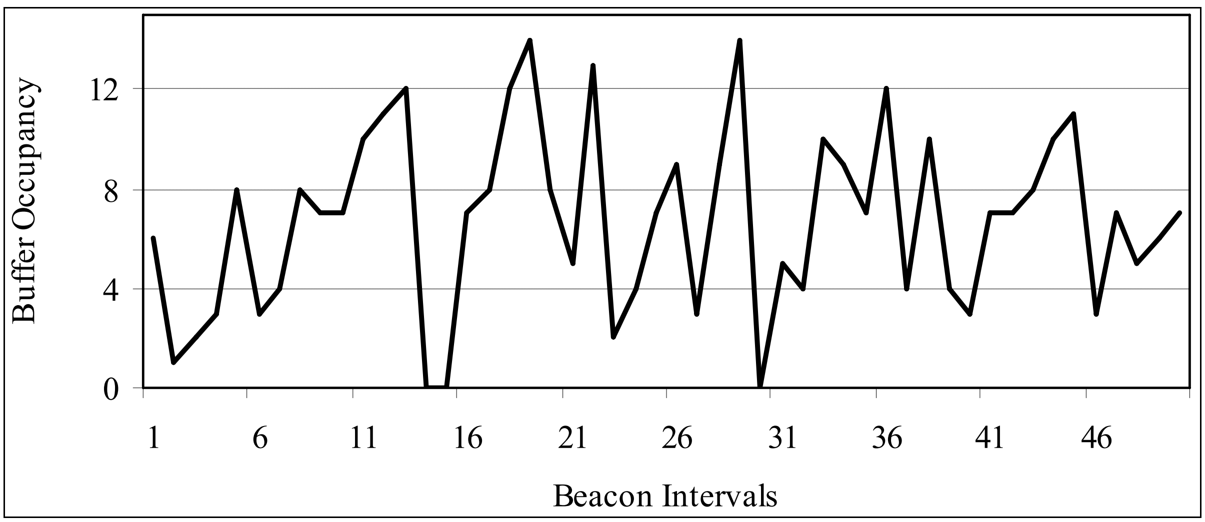

Fig. IV.7. AP buffer occupancy in frames due to deferred transmissions using the proposed PM

Chapter V

3G performance optimization

According to [AGM05], the QoS management has to be defined for some set of indicators, called KPI, Key Performance Indicators, which value is able to provide the health status of the network. In the case of a 3G system (described in Chapter I, Section 1.3) the QoS management (described in Chapter II, Section 2.3) could be improved by introducing a set of KPIs, analysing their impact in the 3G applications, and setting some threshold depending on the service provided. These are the topics of the next sections, ending with a KPI impact analysis in the UMTS domain and some indications on how to collect these measurements.

5.1. KPI: Key Performance Indicators

In the Chapter II the attributes for UMTS QoS Classes were reported, but these attributes cannot be directly used to extract the KPIs since more high-level KPIs have to be defined. Additionally, such attributes describe the QoS for only the UMTS network part while end-to-end service requires the performance analysis also for the rest of the network, which can be for example a wired IP network.

The KPI definition procedure has to be performed for each type of service deployed separately. It means that different quality aspects have to be considered for each different service: voice, video-streaming, web browsing, video-telephony, etc.

In the following, some high-level KPIs are provided. These refer to the accessibility, integrity and retainability performance aspects according to [ITU] as well as to network performance and video streaming service performance.

Accessibility

Accessibility performance can be expressed by means of the following KPI:

- **Call Block Probability (CBP)**: this metric represents the probability that a call required by the user is refused by the network, whatever the problem causes the blocking event (network resource unavailability, wrong routing configuration, etc.). Such a KPI is interesting for the **conversational** and **streaming classes** while is usually not important for the others since the relevant connections are typically always on. The present metric can usually be evaluated at both media-gateway and control level. In the latter case, information about accessibility performance on total traffic served by network domains can be estimated. CBP can be easily mapped into the GoS (Grade of Service) concept used in traditional PSTN networks, except for the fact that the GoS is referred to single trunks while the CBP refers to a path.

Retainability

Retainability performance generally depends on transmission tolerances and on system performance in terms of reliability of the propagation medium. The following measure is accessible for data services:

For dinamically set connections the following KPI can be identified (**conversational** and **streaming classes**):

- **Total number of disconnected active connections in a period T**: active connections are intended to allow for either the case of effective communication between users or the case of "ringing" connections, in other words, all the calls that have successful completed the connection set-up process.

For permanent connections (**interactive** and **background classes**)

- **Permanent connections failure**: this KPI provides information about failures of permanent connections. A failure can be seen for the user viewpoint as an interruption of the service. In other words, this KPI gives a measure of the ability of the network of "retain" data connections.

Integrity

Integrity performance is an indication of the degree to which a service is provided without excessive impairments, once obtained. Actually, temporary inabilities could influence the service, avoiding reaching QoS agreements.

Conversational and **streaming** services put limits on the maximum packet delay per connection in order to guarantee real-time communications. In case of data traffic, an emerging application in IP networks is multimedia data streaming, mainly due to the expansion of Voice over IP, videoconference and audio-video broadcast services. In these scenarios, since no reliable transport protocol is used (e.g., User Datagram Protocol [UDP]), data packet loss ratio has to be considered. Instead, the overall time spent for transmitting a certain amount of data is no longer important being the application time sensitive. Furthermore, data packet transfer delay and data packet transfer delay variation have to be considered. Then, for the conversational and streaming classes the following KPIs are of interest:

- **Packet Transfer Delay**: it is defined as the time elapsed between the departure time of a data packet from the generating end-system and the arrival time at the destination. This value is the accumulation of transfer delay at each node in the path between the two end-systems. It is encountered at a node includes switching delay that is usually negligible, queuing delay that depends on traffic load respect to output throughput, transmission delay (i.e, time to transmit a over the egress link) and propagation delay that it depends on link length and propagation factor
- **Packet Delay Variation**: it is expressed as the difference between maximum and minimum packet transfer delay (i.e, Max-Transfer Delay minus fixed delay components). This KPI allows the evaluation of the maximum possible delay between two consecutive packets deterministically spaced. It also permits the estimation of the worst possible amount of clumping due to queuing.
- **Packet Loss Ratio (PLR)**: losses occur because of overrun of buffering resources due to simultaneous arrivals of bursts from different connections. Connection Admission Control (CAC) scheduling and queuing strategies can affect the amount of packet losses. Packet loss may also occur during failure of components and protection switching. It is defined on a per connection basis as:

  $$PLR = \frac{\text{Lost packets}}{\text{Total transmitted packets}}$$

  where lost packets include either those that do not reach destination or are received corrupted.

Due to their inherent various nature, delay and loss parameters are typical example of KPIs to be extracted by means of external tools.

In case of **interactive** and **background** classes, other performance metrics have to be taken into account to estimate user degree of satisfaction. Thus, it is necessary to simulate network behaviour in case of specific test applications. One of the most important is the file transfer application, which is the main source of traffic in a data network. Over a resilient transport protocol (e.g., TCP) that guarantees correct transmission of data, quality of service in a file transfer session is the overall time spent to complete a transfer. This is directly obtained based on the connection throughput, defined as follows:

- **Data Throughput**: it represents the effective data transfer in the analyzed end-to-end path. Then, throughput consists on the rate at which packets are successfully received, without counting retransmissions triggered by transport protocol stack or duplicate packets at the receiver. Throughput is evaluated at the end-points only, that is, specific performance counters are activated for the input and output connections at the final nodes.

For all the QoS classes other two KPIs are of interest:

- **Time to response**: This is the time that a client has to wait before the start of playback. The initial waiting time can depend on several factors.
- **Bit error-rate in no-lost packets**: Bit-errors in a significant amount of data in a single packet can make multimedia data

Network metrics

Other performance metrics cannot be directly associated to the concept of serveability. Some of them give indication about the global QoS level performed, since they have influence on the end-to-end service provided. This is the case, for instance, of buffer occupancy measures. In case of a rate-based control mechanism (User Policy Control/Network Policy Control active functions), buffer occupancy together with link utilization is an essential metric to evaluate. If at any point in time, the buffer occupancy is very high, this means that the control mechanism is accepting more calls into the network than it should. However, if the link utilization is low when sources are active, this means that the control mechanism is overreacting.

Differently, not including rate-based control mechanisms, buffer occupancy provides only an indication on how the network is able to handle the input traffic. If the buffer is close to the congestion level, some actions should be undertaken (cell/packet discard) or rate-based control mechanism should be activated for the connections that are exceeding the contracted traffic.

Occupancy measures include the following KPIs:

- **Instantaneous Queue Size**: the instantaneous value of the current queue depth, or the packet count of a given connection (in case of per-connection accounting strategy), can be used as an occupancy measure. From this the queue growth can be obtained that provide an useful information about the trends in buffer utilization.

Occupancy affects the overall QoS provided by the system, since it has a direct influence on delay and loss performance. The present KPI is evaluated per service class allowing a separated analysis of the provided QoS per single service supported.

Further KPIs worth to be extracted for different reasons are detailed in the following:

- **Goodput**: it is defined as the ratio between current and maximum achievable throughput, limited by possible bottlenecks in the network or at the source. Usually, goodput is defined as:

  $$Goodput = \frac{\text{packets sent}}{\text{packets retransmitted} + \text{packets sent}}$$

  Packets that need to be resent in order to reach destination are taken into account in order to estimate the efficiency of the connection. For goodput computation it is necessary to know the packet/frame length. This KPI is a clear example of network metric: by the present measure an operator is able to realize the effectiveness of a connection. The QoS delivered to the end user can be not affected, but the network resources are improperly exploited.
- **Link Occupancy**: it is computed as the ratio between sent bytes and link capacity, expressed in terms of packets/sec or bps. This KPI estimates the network utilization level and, therefore the correct exploit of the link budget.
- **Control Processor Load (CP Load)**: it measures control processor load. This metric can be taken into account in order to localize possible network problems in the software configuration or in the call handling functionalities.
- **Routing Failure**: metrics relevant to routing signalling are important to localize possible blocked routed due to wrong protocol configuration. The monitoring of node state for the main routes in the network allows the localization of potential configuration problems.

Metrics for streaming applications

The basis to deal with multimedia streaming media application is the intelligent use of receiver side buffering. The market leading streaming vendors use significant client-side buffering to reduce the effects of jittering. This is quite important to avoid temporary buffer underflow but, on the other hand, it also introduces large start-up delays as the buffer fills. In order to monitor the functioning of such component the following KPI should be used:

- **Decoding buffer overflow**: the monitoring of the number of occurrences of buffer overflow provides a measure of the number packet losses.
- **Decoding buffer underflow**: the decoding buffer underflow would cause skipped frames and frozen display.

- **Errors in frame decoding**: this measures the number of occurrences of bit-errors or system faults that caused the missing of a decoded frame. The type of error is quite different if the errored frame is an intra or inter coded one.
- **Errors in slice decoding**: this measures the number of occurrences of bit-errors or system faults that caused the missing of a slice in a frame. Also for this type of error it important to distinguish the occurrences related to intra and inter frames.
- **Decoding synchronization losses**: depending on the nature of bit-error (long bust errors) it may occur that the decoder completely loses the bitstream decoding synchronization. This requires the decoder to find a recognizable time-stamp for the re-synchronization, losing a certain amount of data and degrading the video quality.

5.2. KPI and their impacts

A typical user is not concerned with how a particular service is provided. However, the user is interested in comparing one service with another in terms of universal, user-oriented performance parameters which apply to any end-to-end service. From a user's perspective, performance should be expressed by parameters which:

- focus on user-perceivable effects, rather than their causes within the network;
- are independent of the networks internal design;
- take into account all aspects of the service from the user's point of view which can be objectively measured at the service access point;
- can be assured to a user by the service providers(s).

With these considerations in mind, this Section examines the requirements of typical end user applications that can be expected.

The KPI definition has to be performed for each type of service deployed separately. It means that different quality aspects have to be considered for each different service indicated in the previous Section; conversational voice, interactive games, e-mail, real time video, etc.

However, it could be helpful to individuate at high level the impact of the performance aspects according to [ITU] as well as to network performance and video streaming service performance. These are shown in the Table V.1 for the UMTS QoS classes only.

Table V.1. Performance impact

	Conversational	Streaming	Interactive	Background
Accessibility	HIGH	MEDIUM	MEDIUM	-
Retainability	HIGH	LOW	MEDIUM	LOW
Integrity	HIGH	HIGH	MEDIUM	LOW
Network metrics	HIGH	HIGH	MEDIUM	LOW
Metrics for streaming applications	HIGH	HIGH	-	-

Accessibility, retainability, network metrics and metrics for streaming applications KPIs can be applied for all services provided over the CS or PS Core Network domain, while some integrity KPI can be applied to only the packet switching context (see Table V.4).

Once defined the impact at top level, it will be analysed which particular KPI has to be considered for each particular service. This is shown from Table V.2 to Table V.6.

Table V.2. Accessibility impact

			Call Block Probability
Conversational	Audio	Conversational voice	X
	Video	Videophone	X
	Data	Telemetry- two-way control	-
	Data	Interactive games	-
	Data	Telnet	-
Streaming	Audio	Speech, mixed speech and music, medium and high quality music	-
	Video	Movie clips, surveillance, real-time video	X
	Data	Bulk data transfer/retrieval, layout and synchronisation information	-
	Data	Still image	-
Interactive	Audio	Voice messaging	-
	Data	Web-browsing - HTML	-
	Data	Transaction services - high priority e.g. e-commerce, ATM	X
	Data	E-mail (server access)	X
Background	Data	Fax	-
	Data	E-mail arrival notification	-
	Data	Low priority transaction service	-

Table V.3. Retainability impact

			Total number of disconnected active connections in a period T	**Permanent connections failure**
Conversational	Audio	Conversational voice	**X**	**-**
	Video	Videophone	**X**	**-**
	Data	Telemetry- two-way control	**-**	**-**
	Data	Interactive games	**X**	**-**
	Data	Telnet	**-**	**-**
Streaming	Audio	Speech, mixed speech and music, medium and high quality music	**X**	**-**
	Video	Movie clips, surveillance, real-time video	**X**	**-**
	Data	Bulk data transfer/retrieval, layout and synchronisation information	**-**	**-**
	Data	Still image	**-**	**-**
Interactive	Audio	Voice messaging	**-**	**-**
	Data	Web-browsing - HTML	**-**	**X**
	Data	Transaction services - high priority e.g. e-commerce, ATM	**-**	**X**
	Data	E-mail (server access)	**-**	**X**
Background	Data	Fax	**-**	**-**
	Data	E-mail arrival notification	**-**	**-**
	Data	Low priority transaction service	**-**	**-**

Table V.4. Integrity impact

			Packet Transfer Delay		**Packet Delay Variation**		**Packet Loss Ratio**		**Data Throughput**		**Time to Response**		**Bit error-rate in no-lost packets**	
			PS	**CS**	**PS**	**CS**	**PS**	**CS**	**PS**	**CS**	**PS**	**CS**	**PS**	**CS**
Conversational	Audio	Conversational voice	**X**	**X**	**X**	-	**X**	**-**	**-**	**-**	**X**	**X**	**X**	**X**
	Video	Videophone	**X**	**X**	**X**	**-**	**X**	**-**	**-**	**-**	**X**	**X**	**X**	**X**
	Data	Telemetry- two-way control	**X**	**X**	**X**	**-**	**X**	**-**	**-**	**-**	**-**	**-**	**X**	**X**
	Data	Interactive games	**X**	**X**	**X**	**-**	**X**	**-**	**-**	**-**	**X**	**X**	**X**	**X**
	Data	Telnet	**X**	**X**	**-**	**-**	**X**	**-**	**-**	**-**	**-**	**-**	**X**	**X**
Streaming	Audio	Speech, mixed speech and music, medium and high quality music	**X**	**X**	**X**	**-**	**X**	**-**	**-**	**-**	**X**	**X**	**X**	**X**
	Video	Movie clips, surveillance, real-time video	**X**	**X**	**X**	**-**	**X**	**-**	**-**	**-**	**X**	**X**	**X**	**X**
	Data	Bulk data transfer/retrieval, layout and synchronisation information	**X**	**X**	**X**	**-**	**X**	**-**	**-**	**-**	**-**	**-**	**X**	**X**
	Data	Still image	**-**	**-**	**-**	**-**	**X**	**-**	**-**	**-**	**-**	**-**	**X**	**X**

			Packet Transfer Delay		**Packet Delay Variation**		**Packet Loss Ratio**		**Data Throughput**		**Time to Response**		**Bit error-rate in no-lost packets**	
			PS	**CS**	**PS**	**CS**	**PS**	**CS**	**PS**	**CS**	**PS**	**CS**	**PS**	**CS**
Interactive	Audio	Voice messaging	**-**	**-**	**X**	**-**	**-**	**-**	**X**	**X**	**-**	**-**	**-**	**-**
	Data	Web-browsing - HTML	**-**	**-**	**-**	**-**	**-**	**-**	**X**	**X**	**-**	**-**	**-**	**-**
	Data	Transaction services - high priority e.g. e-commerce, ATM	**X**	**X**	**-**	**-**	**X**	**-**	**X**	**X**	**X**	**X**	**X**	**X**
	Data	E-mail (server access)	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**X**	**X**
Background	Data	Fax	**-**	**-**	**-**	**-**	**-**	**-**	**X**	**X**	**-**	**-**	**-**	**-**
	Data	E-mail arrival notification	**-**	**-**	**-**	**-**	**-**	**-**	**X**	**X**	**-**	**-**	**-**	**-**
	Data	Low priority transaction service	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**-**	**X**	**X**

Table V.5. Network metrics impact

			Istantaneous queue size	**Goodput**	**Link occupancy**	**CP load**	**Routing failure**
Conversational	Audio	Conversational voice	**X**	**X**	**-**	**X**	**X**
	Video	Videophone	**X**	**X**	**X**	**X**	**X**
	Data	Telemetry- two-way control	**-**	**X**	**-**	**-**	**-**
	Data	Interactive games	**X**	**X**	**X**	**X**	**X**
	Data	Telnet	**-**	**-**	**-**	**-**	**-**
Streaming	Audio	Speech, mixed speech and music, medium and high quality music	**X**	**X**	**X**	**X**	**X**
	Video	Movie clips, surveillance, real-time video	**X**	**X**	**X**	**X**	**X**
	Data	Bulk data transfer/retrieval, layout and synchronisation information	**X**	**-**	**X**	**X**	**-**
	Data	Still image	**-**	**X**	**X**	**X**	**-**
Interactive	Audio	Voice messaging	**X**	**-**	**-**	**-**	**-**
	Data	Web-browsing - HTML	**-**	**X**	**-**	**-**	**-**
	Data	Transaction services - high priority e.g. e-commerce, ATM	**-**	**X**	**-**	**-**	**X**
	Data	E-mail (server access)	**-**	**-**	**-**	**-**	**-**
Background	Data	Fax	**-**	**-**	**-**	**-**	**X**
	Data	E-mail arrival notification	**-**	**-**	**-**	**-**	**-**
	Data	Low priority transaction service	**-**	**-**	**-**	**-**	**X**

Table V.6. Streaming applications metrics impact

			Decoding buffer overflow	Decoding buffer underflow	Errors in frame decoding	Errors in slice decoding	Decoding synchronization losses
Conversational	Audio	Conversational voice	X	X	X	X	X
	Video	Videophone	X	X	X	X	X
	Data	Telemetry- two-way control	-	-	-	-	-
	Data	Interactive games	X	X	X	X	X
	Data	Telnet	-	-	-	-	-
Streaming	Audio	Speech, mixed speech and music, medium and high quality music	X	X	X	X	X
	Video	Movie clips, surveillance, real-time video	X	X	X	X	X
	Data	Bulk data transfer/retrieval, layout and synchronisation information	-	-	-	-	-
	Data	Still image	-	-	-	-	-
Interactive	Audio	Voice messaging	-	-	-	-	-
	Data	Web-browsing - HTML	-	-	-	-	-
	Data	Transaction services - high priority e.g. e-commerce, ATM	-	-	-	-	-
	Data	E-mail (server access)	-	-	-	-	-
Background	Data	Fax	-	-	-	-	-
	Data	E-mail arrival notification	-	-	-	-	-
	Data	Low priority transaction service	-	-	-	-	-

5.3. KPI thresholds

This Section outlines the QoS requirements that shall be provided to the end user / applications and describes them as requirements between communicating entities (i.e. end to end). The QoS values in the tables represent end to end performance, including mobile to mobile calls and satellite components. Delay values represent one-way delay (i.e. from originating entity to terminating entity). The values included in the following tables are commonly accepted values from an end-user viewpoint. The delay contribution within the mobile network should be kept to minimum since there may be additional delay contributions from external networks.

The following tables from Table V.7 to Table V.9, [3G221], further elaborate end user / application QoS requirements, even though the thresholds are provided for a subset of the KPI related to the integrity only.

Obviously, these parameters have to be looser than the attributes previous specified for the UMTS bearer services (see also [3G231]) .

Table V.7. End-user Performance Expectations - Conversational / Real-time Services

Medium	Application	Degree of symmetry	Data rate	Key performance parameters and target values		
				Packet Transfer Delay (End-to-end One-way)	**Packet Delay Variation** (within a call)	**Packet Loss Ratio** (frame error ratio)
Audio	Conversational voice	Two-way	4-25 kb/s	<150 msec preferred <400 msec limit Note 1	< 1 msec	< 3% FER
Video	Videophone	Two-way	32-384 kb/s	< 150 msec preferred <400 msec limit Lip-synch : < 100 msec		< 1% FER
Data	Telemetry - two-way control	Two-way	<28.8 kb/s	< 250 msec	N.A	Zero
Data	Interactive games	Two-way	< 1 KB	< 250 msec	N.A	Zero
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 250 msec	N.A	Zero

Note 1: The overall one way delay in the mobile network (from UE to PLMN border) is approximately 100msec.

Table V.8. End-user Performance Expectations - Streaming Services

Medium	Application	Degree of symmetry	Data rate	Key performance parameters and target values		
				Packet Transfer Delay (Start-up)	**Packet Delay Variation**	**Packet Loss Ratio** (at session layer)
Audio	Speech, mixed speech and music, medium and high quality music	Primarily one-way	5-128 kb/s	< 10 sec	< 2sec	< 1% Packet loss ratio
Video	Movie clips, surveillance, real-time video	Primarily one-way	20-384 kb/s	< 10 sec	<2 sec	< 2% Packet loss ratio
Data	Bulk data transfer/retrieval , layout and synchronisation information	Primarily one-way	< 384 kb/s	< 10 sec	N.A	Zero
Data	Still image	Primarily one-way		< 10 sec	N.A	Zero

Table V.9. End-user Performance Expectations - Interactive Services

Medium	Application	Degree of symmetry	Data rate	Key performance parameters and target values		
				Packet Transfer Delay (One-way)	**Packet Delay Variation**	**Packet Loss Ratio** (frame error ratio)
Audio	Voice messaging	Primarily one-way	4-13 kb/s	< 1 sec for playback < 2 sec for record	< 1 msec	< 3% FER
Data	Web-browsing - HTML	Primarily one-way		< 4 sec /page	N.A	Zero
Data	Transaction services – high priority e.g. e-commerce, ATM	Two-way		< 4 sec	N.A	Zero

Performance requirements for conversational real-time

The real time conversation scheme is characterised by the transfer time that shall be low because of the conversational nature of the scheme and at the same time that the time relation (variation) between information entities of the stream shall be preserved in the same way as for real time streams. The maximum transfer delay is given by the human perception of video and audio conversation. Therefore the limit for acceptable transfer delay is very strict, as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is therefore both significantly lower and more stringent than the round trip delay of the interactive traffic case.

Real time conversation - fundamental characteristics for QoS:

- preserve time relation (variation) between information entities of the stream;
- conversational pattern (stringent and low delay).

The resulting overall requirement for this communication scheme is to support conversational real time services with low transfer delay as given by the human perception (there are less hard requirements on packet loss ratio).

A real-time streaming application is one that delivers time-based information in real-time, where time-based information is user data that has an intrinsic time component. Video, audio and animation are examples of time-based information, in that they consist of a continuous sequence of data blocks that shall be presented to the user in the right sequence at pre-determined instants.

Conversational voice

Audio transfer delay requirements depends on the level of interactivity of the end users. To preclude difficulties related to the dynamics of voice communications, ITU-T Recommendation

G.114 recommends the following general limits for one-way transmission time (assuming echo control already taken care of):

- 0 to 150 ms: referred range [<30ms, user does not notice any delay at all, <100ms, user does not notice delay if echo cancellation is provided and there are no distortions on the link]
- 150 to 400 ms: acceptable range (but with increasing degradation)
- above 400 ms: unacceptable range

The human ear is highly intolerant of short-term delay variation (jitter) it is therefore paramount that this is reduced as lower level as is practical. A limit as low as 1 msec is suggested as a target.

Requirements for information loss are influenced by the fact that the human ear is tolerant to a certain amount of distortion of a speech signal. It is has been suggested in studies that acceptable performance is typically obtained with frame erasure rates (FER) up to 3 %.

A connection for a conversation normally requires the allocation of symmetrical communication resources, with the average hold time of a call being in the region of 2 minutes.

Videophone

Videophone implies a full-duplex system, carrying both video and audio and intended for use in a conversational environment. As such, in principle the same delay requirements as for conversational voice will apply, i.e. no echo and minimal effect on conversational dynamics, with the added requirement that the audio and video must be synchronised within certain limits to provide “lip-synch” (i.e. synchronisation of the speaker’s lips with the words being heard by the end user). In fact, due to the long delays incurred in even the latest video codecs, it will be difficult to meet these requirements.

Once again, the human eye is tolerant to some loss of information, so that some degree of packet loss is acceptable depending on the specific video coder and amount of error protection used. It is expected that the latest video codecs will provide acceptable video quality with frame erasure rates up to about 1%.

Interactive games

Requirements for interactive games are obviously very dependent on the specific game, but it is clear that demanding applications will require very short delays, and a value of 250 msecs is proposed, consistent with demanding interactive applications.

Two-way control telemetry

Two-way control telemetry is included here as an example of a data service which does require a real-time streaming performance. Clearly, two-way control implies very tight limits on allowable delay and a value of 250 msec is proposed, but a key differentiator from the voice and video services in this category is the zero tolerance for information loss (obvious if you are controlling an important industrial process, for example).

Telnet

Telnet is included here with a requirement for a short delay in order to provide essentially instantaneous character echo-back.

Performance requirements for streaming services

This scheme is one of the newcomers in data communication, raising a number of new requirements in both telecommunication and data communication systems. First of all it is a mainly unidirectional stream with high continuous utilisation (i.e. having few idle/silent periods.) It is also characterised by that the time relations (variation) between information entities (i.e. samples, packets) within a flow must be preserved, although it does not have any requirements on low transfer delay.

The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. But as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application. Acceptable delay variation is thus much greater than the delay variation given by the limits of human perception.

Real time streams - fundamental characteristics for QoS:

- unidirectional continuous stream;
- preserve time relation (variation) between information entities of the stream.

The resulting overall requirement for this communication scheme is to support streaming real time services having unidirectional data flows with continuous utilisation. Note that there are less stringent requirements on delay and packet loss ratio, i.e. the ratio of lost or corrupted packets out of all packets sent.

Audio streaming

Audio streaming is expected to provide better quality than conventional telephony, and requirements for information loss in terms of packet loss will be correspondingly tighter. However, as with voice messaging, there is no conversational element involved and delay requirements can be relaxed, even more so than for voice-messaging.

One-way video

The main distinguishing feature of one-way video is that there is no conversational element involved, meaning that the delay requirement will not be so stringent, and can follow that of streaming audio.

Bulk data

This category includes file transfers, and is clearly influenced by the size of the file. As long as there is an indication that the file transfer is proceeding, it is reasonable to assume some what longer tolerance to delay than for a single Web-page.

Still image

This category includes a variety of encoding formats, some of which may be tolerant to information loss since they will be viewed by a human eye. However, given that even single bit errors can cause large disturbances in other still image formats, it is argued that this category should in general have zero information loss. However, delay requirements for still image transfer are not stringent, given that the image tends to be built up as it is being received, which provides an indication that data transfer is proceeding.

Telemetry (monitoring)

Monitoring covers a wide range of applications, but in this category it is taken to apply to relatively low priority activities, e.g. status updating, rather than control.

Performance requirements for Interactive Services

Interactive traffic is the other classical data communication scheme that on an overall level is characterised by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round trip delay time is therefore one of the key attributes. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

Interactive traffic - fundamental characteristics for QoS:

- request response pattern;
- preserve payload content.

The resulting overall requirement for this communication scheme is to support interactive non-real time services with low round-trip delay.

Voice messaging and dictation

Requirements for information loss are essentially the same as for conversational voice, but a key difference here is that there is more tolerance for delay since there is no direct conversation involved. The main issue, therefore becomes one of how much delay can be tolerated between the user issuing a command to replay a voice message and the actual start of the audio. There is no precise data on this, but a delay of the order of a few seconds appears reasonable for this application.

Data

Although there may be some exceptions, as a general rule it is assumed that from a user point of view, a prime requirement for any data transfer application is to guarantee essentially zero loss of information. At the same time, delay variation is not applicable. The different applications therefore tend to distinguish themselves on the basis of the delay which can be tolerated by the end-user from the time the source content is requested until it is presented to the user.

Web-browsing

This category is referred to retrieving and viewing the HTML component of a Web page, other components e.g. images, audio/video clips are dealt with under their separate categories. From the user point of view, the main performance factor is how fast a page appears after it has been requested. A value of 2-4 seconds per page is proposed, however improvement on these figures to a target figure of 0.5 seconds wound be desirable.

High-priority transaction services (E-commerce)

The main performance requirement here is to provide a sense of immediacy to the user that the transaction is proceeding smoothly. A value of 2-4 seconds is suggested to be acceptable to most users.

E-mail (server access)

E-mail is generally thought to be a store and forward service which in principle can tolerate delays of several minutes or even hours. However, it is important to differentiate between communications between the user and the local email server and server to server transfer. When the user communicates with the local mail server, there is an expectation that the mail will be transferred quite rapidly, although not necessarily instantaneously. Consistent with the research findings on delay tolerance for Web-browsing, a requirement of 2-4 seconds is proposed.

Performance requirements for Background applications

Background traffic is one of the classical data communication schemes that on an overall level is characterised by that the destination is not expecting the data within a certain time. The scheme is thus more or less delivery time insensitive. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

Background traffic - fundamental characteristics for QoS:

- the destination is not expecting the data within a certain time;
- preserve payload content.

The resulting overall requirement for this communication scheme is to support non-real time services without any special requirement on delay.

A background application is one that does not carry delay information. In principle, the only requirement for applications in this category is that information should be delivered to the user essentially error free. However, there is still a delay constraint, since data is effectively useless if it is received too late for any practical purpose.

Fax

Fax is included in this category since it is not normally intended to be an accompaniment to real-time communication. Nevertheless, there is an expectation in most business scenarios that a fax will be received within about 30 seconds. The information loss requirement is based on established wireline requirements for a Group 3 fax. As for the symmetry this should provide the required through put in the sending direction and the control signalling in backwards direction, hence an asymmetric connection is required.

Email (server to server)

This category is included for completeness, since as mentioned earlier, the prime interest in email is in the access time. There is a wide spread in user expectation, with a median value of several hours.

Low priority transaction services

An example in this category is Short Message Service (SMS). 30 seconds is proposed as an acceptable delivery delay value.

5.4. Network localisation

In this Section, with reference to some services, the impact of network components behaviour to the mentioned end-to-end KPI are described. In particular, the "key" services for the 3G networks are considered, as follows:

- conversational voice;
- SMS/MMS;
- videophone;
- interactive games;
- web-browsing – HTML.

The USIM and Mobile Equipment domains are considered as a single domain, then the network impact will be analysed in three domains:

- user equipment: this domain encompasses a variety of equipment types with different levels of functionality; these equipment types are referred to as user equipment (terminals), and they may also be compatible with one or more existing access (fixed or radio) interfaces e.g. dual mode UMTS-GSM user equipment. The User Equipment is a device allowing a user access to network services. The interface between the UE and the network is the radio interface.
- access network: it consists of the physical entities which manage the resources of the access network and provides the user with a mechanism to access the core network domain.
- core network: it consists of the physical entities which provide support for the network features and telecommunication services. The support provided includes functionality such as the management of user location information, control of

network features and services, the transfer (switching and transmission) mechanisms for signalling and for user generated information.

Table V.10. UMTS domains impact on KPI

	Conversational voice	SMS/MMS	Videophone	Interactive games	Web-browsing - HTML
Accessibility	User equipment / Access network	Core network	User equipment / Access network	Access network / Core network	Core network
Retainability	User equipment / Access network	User equipment / Access network / Core network	User equipment / Access network / Core network	User equipment / Access network / Core network	User equipment / Access network / Core network
Integrity	User equipment / Access network / Core network	User equipment / Access network	User equipment / Access network	User equipment / Access network	User equipment / Access network
Network metrics	Core network	Core network	Core network	Core network	Core network
Metrics for streaming applications	Access network / Core network	Access network / Core network	Access network / Core network	Access network / Core network	Access network / Core network

5.5. KPI Extraction

Network services are differently managed and it is necessary to study, propose and develop what is needed in order to ensure the consistence of a given end-to-end service.

Several solutions are available (passive, active). These tools should be analyzed and investigated in order to determine which technique (passive, active) is better suited for a given service or a given indicator. Other tools will be also taken into account in order to "track" the route/path in a domain or between different domains.

Active Measurement. Probe packets for measurement are generated and sent, and the traffic is measured based on the information carried by these packets. This active measurement can check the end-to-end availability, delay and throughput.

Passive Measurement. The traffic passing through the locations where probes are installed is observed. By this method the number of packets, the number of bytes, and packet loss count can be measured and the effective throughput and packet loss probability can be easily calculated. More complex passive measurement tools can also provide information about transfer delay average and variations by tracing the path of real data packets. This requires the two measurement points be synchronized with the same clock.

A preliminary study suggests to adopt passive measurements, even through the use of SNMP (Simple Network Management Protocol) protocol agents. This choice should allow the operator to activate on its machine some counters and then extract the data. These counters have to be studied in order to be correlated to the KPI. For example, the call block probability for the conversational voice should be calculated after extracting the successfully and unsuccessfully calls in the “intelligent network part”.

The methodology to be performed start by the activation of some counters, SNMP external agents and measurement tools. After a period for data collection (audit phase), the KPI has to be calculated basing on the audit.

5.6. Future improvements

Herein, requirements for future work are given:

- As described in Section 5.3, “KPI threshold”, performance requirements for some services have been already defined while for others the relevant thresholds have not been defined yet. This is a task to be accomplished.
- This research have presented a first rough definition of the impact of the network components behaviour to the end-to-end perception of the service quality. This is an import factor to drive network troubleshooting. In particular, a better definition of the key network components, functioning and impact on the different performance aspect is required.
- Once the relationships KPI/network components have been defined, the specific measurements (active or passive) needed to assess the component functioning need to be defined.
- Network troubleshooting can be supported by the use of a simulation tool fed with real traffic measurements obtained from the working 3G network. This would allow the network administrator to predict network faults, malfunctioning and to validate control actions before introducing these in the real network.

Appendix A

Bibliography

[3G221] 3GPP TS 22.105 v6.1.0. 3rd Generation Partnership Project; Technical Specification Group Services and System Aspects Service aspects; *Services and Service Capabilities*, 2003.

[3G230] 3GPP TS 23.002 v5.10.0. 3rd Generation Partnership Project; Technical Specification Group Services and Systems Aspects; *Network architecture*, 2003.

[3G231] 3GPP TS 23.107 v5.8.0. 3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; *Quality of Service (QoS) concept and architecture,* 2003.

[3G238] 3GPP TS 23.821 v1.0.1. 3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; *Architecture Principles for Release 2000,* 2000.

[3G2310] 3GPP TS 23.101 v4.0.0. 3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; *General UMTS Architecture,* 2001.

[AGM05] L. Atzori, D.D. Giusto and M. Murroni "Chapter: QoS Management in Multiservice Packet Networks", *IEC Annual Review of Communications*, vol. 58, 2005

[AR02] C.W. Ahn and R.S. Ramakrishna, "A genetic algorithm for shortest path routing problem and the sizing of population", *IEEE Transaction on evolutionary computation*, vol. 6, no. 6, Dec. 2002.

[AR04] L. Atzori and A. Raccis, "Network Capacity Assignment for Multicast Services Using Genetic Algorithms", *IEEE Communications Letters*, vol. 8, no. 6, pp. 403-405, June 2004.

[BB00] D. Bienstock and A. Bley, "Capacitated network design with multicast commodities", ZIB-report 00-14, Konrad-Zuse-Zentrum four informationstechnik Berlin, 2000.

[BCD01] L. Bononi, M. Conti and L. Donatiello, "A distributed Mechanism for Power Saving in IEEE 802.11 Wireless LANs, " *ACM Mobile Networks and Applications (ACM MONET) Journal*, Vol 6, 2001, pp 211-222.

[BVSN05] R. Bhattacharya, P. Venkateswaran, S.K. Sanyal, R. Nandi, , "Genetic algorithm based efficient routing scheme for multicast networks", *IEEE International*

Conference on Personal Wireless Communications (ICPWC 2005), pp. 500 - 504, Jan 2005.

[CGY00] S. Chen, O. Günlük, and B. Yener, "The multicast packing problem", *IEEE/ACM Transactions on Networking*, vol. 8, pp. 311-318, June 2000.

[CH04] H. Chen, C.-W. Huang, "Power Management Modeling and Optimal Policy for IEEE 802.11 WLAN Systems", *Proc. IEEE Vehicular Technology Conference*, 2004. VTC2004-Fall, Vol. 6, September 2004, pp. 4416-4421.

[CHH97] C. Chung, S. Hong, and H. Huh, "A fast multicast routing algorithm for delay-sensitive applications", *Proceedings of Globecom'97*, pp. 1898-1902, 1997.

[CSE04] Y. Chen, N. Smavatkul, and S. Emeott, "Power Management for VoIP over IEEE 802.11 WLAN", *Proc. IEEE Wireless Communications and Networking Conference (WCNC)*, Vol. 3, 21-25 March 2004, pp. 1648-1653.

[CSI01] Cisco Systems Inc., "OSPF design guide", 2001.

[CYX04] L. Chen, Z. Yang, Z. Xu, "A degree-delay-constrained genetic algorithm for multicast routing tree", *The Fourth International Conference on Computer and Information Technology* (CIT), pp. 1033 - 1038, September 2004.

[D59] E.W. Dijkstra, "A note on two problems in connection with graphs", *Numerische Mathematik* 1, pp. 269-271, 1959.

[D91] L. Davis, *Handbook of Genetic Algorithms*. Van Nostrand Reinhold, 1991.

[D95] T. Dawn, "Nature Shows the Way to Discover Better Answers", *Scientific Computing World 6*, pp. 23-27, 1995.

[ERP01] M. Ericsson, M.G.C. Resende, and P.M. Pardalos. "A genetic algorithm for the weight setting problem in OSPF routing" Technical report, AT&T Labs Research, October 2001.

[FT00] B. Fortz and M. Thorup, "Internet traffc engineering by optimizing OSPF weights" *Proceedings of IEEE INFOCOM'2000*, pp. 519-528, 2000.

[FZ02] C. H. Foh, M. Zukerman, "A New Technique for Performance Evaluation of Random Access Protocols", *Proc. IEEE International Conference on Communications*, ICC 2002, Vol. 4, 28 April-2 May 2002, pp. 2284-2288.

[G89] D.E. Goldberg, *Genetic Algorithms in Search, Optimization & Machine Learning*. Addison-Wesley, 1989.

[H92] J. H. Holland. *Adaptation in Natural and Artificial Systems*. The University of Michigan Press, Second edition, 1992.

[H94] J. Heitkoetter and D. Beasley. *The Hitch-Hiker's Guide to Evolutionary Computation: A List of Frequently Asked Questions (FAQ)*. USENET: comp.ai.genetic, 1994.

[HDY00] R.H. Hwang, W.Y. Do, and S.C. Yang, "Multicast Routing Based on Genetic Algorithms", *Journal of information science and engineering*, vo1.16, pp.885-901, 2000.

[HE04] M. Hamdan and M.E. El-Hawary, "Multicast routing with delay and delay variation constraints using genetic algorithm", *Canadian Conference on Electrical and Computer Engineering*, vol. 4, pp. 2363 - 2366, May 2004.

[HFDMG04] A.T. Haghighat, K. Faez, M. Dehghan, A. Mowlaei and Y. Ghahremani, "GA-based heuristic algorithms for bandwidth-delay-constrained least cost multicast routing", *Computer Communications*, vol. 27, pp. 111-127, 2004.

[HY01] K. Holmberg and D. Yuan. "Optimization of internet protocol network design and routing". Technical Manuscript, Linkoping University, *Networks*, 2001.

[ITU] ITU-T Rec. E 800, "Telephone Network and ISDN Quality of Service, Network Management and Traffic Engineering", SG 2, Geneva, Switzerland, August 1994

[J95] X. Jia, "A distributed algorithm of delay-bounded multicast routing for multimedia applications in wide area networks", *IEEE/ACM Transaction on Networking*, vol. 6, pp. 828-837, 1995.

[JM01] M. Jones and D. Mitchell, "JUNOS Internet Software Con Guide: Routing and Routing Protocols, Release 4.3", Juniper Networks Inc., Sunnyvale, CA, USA, 2001.

[JV02] E.-S. Jung and N. Vaidya, "An Energy Efficient MAC Protocol for Wireless LANs", in *Proc. IEEE INFOCOM 2002*, Vol. 1, June 2002.

[L01] C. Long, *IP Network Design*. Osborne/McGraw-Hill, 2001.

[LN05] H. Lei, A.A. Nilsson, "A Power Management Scheme for the IEEE 802.11 Based WLANs", *Proc. IEEE International Conference on Performance, Computing and Communications, IPCCC 2005*, April 2005, pp. 123-127.

[LW93] F.Y.S. Lin and J.L. Wang, "Minimax open shortest path rst routing algorithms in networks networks supporting the SMDS service", Technical Program, Conference Record, *IEEE International Conference on Communications*, volume 2, pp. 666-670, 1993.

[LY98] Y. W. Leung and B. T. Yang, "*Lower bound for multimedia multicast routing*", IEE Proc. in Communication, Vol. 145, pp. 87-90, 1998.

[M98] J. Moy, "OSPF version 2" Request for Comments 2328, Internet Engineering Task Force, April 1998.

[PAS95] V. Priwan, H. Aida, and T. Saito "The multicast tree based routing for the complete broadcast multipoint-to-multipoint communications", *IEICE Transactions of the Communications*, vol. E78-B, pp.720-728, 1995.

[PF01] M. Prytz and A. Forsgren, "Dimensioning of multicast network that uses shortest path routing distribution trees", Report TRITA-MAT-2001-OS2, Department of Mathematics, Royal Institute of Technology, 2001.

[RVC01] E. Rosen, A. Viswanathan, and R. Callon, "Multiprotocol label switching architecture", *Request for Comments 3031, Internet Engineering Task Force*, January 2001.

[S98] W. Stallings, "*High-Speed Networks, TCP/IP and ATM Design Principles*", Prentice Hall, 1998.

[SKK00] D. Staehle, S. Kohler, and U. Kohlhaas, "Towards an optimization of the routing parameters for IP networks" *Research report TR 258, Department of Computer Science*, University of Wourzburg, 2000.

[SLS96] A. Shaikh, S. Lu, and K. Shin, "Localized multicast routing", *Proceedings of IEEE Globecom'95*, pp. 1352-1356, 1996.

[SR02] L. Sanna Randaccio, "*Performance review for multi-service switches*", M.Sc. dissertation, Dept. Electronic Engineering, Univ. Cagliari, Italy, 2002.

[STV01] H. Sayoud, K. Takahashi, and B. Vaillant, "Designing communication networks topologies using steady-state genetic algorithms", *IEEE Communication Letters*, vol. 5, pp. 113-115, March 2001.

[T95] M. Tomassini, A survey of genetic algorithms. In D.Stauffer, editor, *Annual Reviews of Computational Physics*, volume III, pages 87-118. World Scientific, 1995. Also available as: Technical Report 95/137, Department of Computer Science, Swiss Federal Institute of Technology, Lausanne, Switzerland, July, 1995.

[T96] M. Tomassini, Evolutionary algorithms. In E. Sanchez and M. Tomassini, editors, *Towards Evolvable Hardware*, volume 1062 of *Lecture Notes in Computer Science*, pages 19-47. Springer-Verlag, Berlin, 1996.

[T00] E. S. C. Takahashi, "Application aware scheduling for power management on IEEE 802.11", *Proc. IEEE International Conference on Performance Computing and Communications*, IPCCC 2000, February 2000, pp. 247-253.

[TC01] Y.C. Tay, K.C. Chua, "A capacity Analysis for the IEEE 802.11 MAC Protocol", *Wireless Networks* Vol. 7, 2001, pp 159-171.

[TH00] D. Thaler and C. Hopps, "Multipath issues in unicast and multicast next-hop selection", Request for Comments 2991, Internet Engineering Task Force, November 2000.

[THH02] Y. C. Tseng, C. S. Hsu and T. Y. Hsieh, "Power-Saving Protocols for IEEE 802.11-Based Multi-Hop Ad Hoc Networks", *Proc. IEEE INFOCOM 2002*, Vol. 1, June 2002, pp. 200-209.

[TM80] H. Takahashi and A. Matsuyama, "An approximation solution for the Steiner tree problem in graphs", *Mathematica Japonica*, vol. 24, pp. 573-577, 1980.

[W88] B. Waxman, "Routing of multipoint connections", *IEEE Journal on Selected Areas in Communications*, vol. 6, pp. 1617-1622, December 1988.

[W00] B. Williamson, "*Developing IP Multicast Networks: The De Guide to Designing and ting Deploying CISCO IP Multicast Networks*", Cisco Press, 2000.

[WA04] X. Wang, D.P. Agrawal, "Analysis and optimization of energy efficiency in 802.11 distributed coordination function", in *Proc. IEEE International Conference on Performance Computing and Communications*, IPCCC 2004, April 2004, pp. 707-712.

[WFB04] A. Weissel, M. Faerber and F. Bellosa, "Application Characterization for Wireless Network Power Management", *Proc. Of the International Conference on Architecture of Computing Systems*, ARCS 2004, March 2004, pp. 231-245.

[Wi99] IEEE Computer Society. "IEEE standard 802.11: wireless LAN medium access control (MAC) and physical layer (PHY) specifications", *The Institute of Electrical and Electronics Engineers*, New York, 1999.

[WLJ99] C. Wang, B. Lai, R. Jan, "Optimum multicast of multimedia streams", *Computer & Operations Research*, vol. 26, pp. 461-480, 1999.

[WLJ02] C. Wang, C. Liang, and R. Jan, "Heuristic algorithms for packing of multiple-group multicasting", *Computer & Operations Research*, vol. 29, pp. 905-924, 2002.

[ZHJL03] J. Zhao, H. Hassanein, J. Wu and J. Luo, "CRMA: A Cycle-breaking Multicast Routing Algorithm for Supporting QoS over the Internet", *Proceedings of the 36th Hawaii International Conference on System Sciences*, 2003.

[ZPG95] Q. Zhu, M. Parsa, and J. Garcia-Luna-Aceves, "A source-based algorithm for delay-constrained minimum-cost multicasting", *Proceedings of IEEE Infocom'95*, pp. 377-385, 1995.

Appendix B

Acronyms table

3G	Third Generation
3GPP	Third-Generation Partnership Project
AC	Authentication Center
ACK	ACKnowledgement
AID	Association IDentifier
AMR	Adaptive MultiRate
ANSI	American National Standards Institute
AP	Access Point
ARIB/TTC	Association of Radio Industries and Business/Telecommunication Technology Committee
ATIM	Announcement TIM
ATM	Asynchronous Transfer Mode
BPSK	Binary Phase Shift Keying
BSC	Base Station Controller
BSS	Basic Service Set *(wireless networks)*
BSS	Base Station Subsystem *(mobile networks)*
BTS	Base Transceiver Station
CAC	Connection Admission Control
CAMEL	Customized Application for Mobile Enhanced Logic
CAP	CAMEL Application Part
CBP	Call Block Probability
CBT	Core Based Trees
CCK	Complementary Code Keying
CCS7	Common Control SS7
CN	Core Network
CODEC	enhanced speech COmpression/DECompression
CP	Control Processor
CRC	Cyclic Redundancy Checking
CRNC	Controlling RNC
CS	Circuit-Switched domain
CSE	CAMEL Service Environment
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
CTS	Clear to Send
CWTS	Chinese Wireless Telecommunication Standard
D–AMPS	Digital Advanced Mobile Phone System
DCF	Distributed Coordination Function
DIFS	Distributed Interframe Space
DL	DownLink
DRNC	Drift RNC
DS	Distribution System
DSS	Distribution System Services

DSSS	Direct Sequence Spread Spectrum
DTIM	Delivery TIM
DVMRP	Distance Vector Multicast Routing Protocol
EDGE	Enhanced Data rates for GSM Evolution
EFR	Enhanced Full Rate
EIFS	Extended interframe space
EIR	Equipment Identity Register
ESS	Extended Service Set
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission USA
FDD	Frequency Division Duplex
FDDI	Fiber Distributed Data Interface
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FER	Frame Error Rate
FHSS	Frequency Hopping Spread Spectrum
FTP	File Transfer Protocol
GA	Genetic Algorithm
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GoS	Grade of Service
GSM	Global System for Mobile Communications
HLR	Home Location Register
HSCSD	High-Speed Circuit-Switched Data
HTML	HyperText Markup Language
IBSS	Independent Basic Service Set
iBSS	infrastructure BSS
ID	IDentity
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IGMP	Internet Group Membership Protocol
IGP	Interior Gateway Protocol
IMT	International Mobile Telecommunications
IP	Internet Protocol
IS–95	Interim Standard ’95
ISDN	Integrated Services Digital Network
IS-IS	Intermediate System - Intermediate System
ISM	Industrial, Scientific, Medical
ISO	International Organization for Standardization
ITU	International Telecommunications Union
IWF	InterWorking Function
KPI	Key Performance Indicator
LAN	Local Area Network
LB	Lower Bound
LLC	Logical Link Control layer
MAC	Medium Access Control
MANET	Multi Hop Ad Hoc Networks
MBONE	Multicast BackBONE

MC	MultiCarrier
MMS	Multimedia Messaging Service
MOSPF	Multicast Open Shortest Path First
MPLS	Multiprotocol Label Switching
MS	Mobile Station
MSC	Mobile services Switching Center
MSDU	MAC Service Data Unit
MSS	Mobile Satellite System
MT	Mobile Termination
N.A.	Not Available
NAV	Network Allocation Vector
NSS	Network and Switching Subsystem
O&M	Operation and Maintenance
OAM	Operation, Administration, and Maintenance
OFDM	Orthogonal Frequency Division Multiplexing
OMC	Operation and Maintenance Center
OSPF	Open Shortest Path First
OSS	Operations Support System
PBCC	Packet Binary Convolutional Coding
PCF	Point Coordination Function
PDN	Packet Data Network
PIFS	PCF interframe space
PIM	Protocol-Independent Multicast
PLMN	Public Land Mobile Network
PLR	Packet Loss Ratio
PM	Power Management
PS	Packet-Switched domain *(mobile networks)*
PS	Power-Saving mode *(wireless networks)*
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAM	Random Access Memory
RF	Radio Frequence
RFC	Request For Comment
RIP	Routing Information Protocol
RNC	Radio Network Controller
RNS	Radio Network System
RP	Rendez-vous Point
RPF	Reverse Path Forwarding
RRM	Radio Resource Management
RTS	Request to Send
SDO	Standards Developing Organization
SGSN	Serving GPRS Support Node
SIFS	Short interframe space
SMS	Short Message Service
SNMP	Simple Network Management Protocol
SPT	Shortest Path Trees
SRNC	Serving RNC
SRNS	Serving Radio Network Subsystem

SS	Station Services
SS7	Signaling System 7
TC	TransCoder
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TD–SCDMA	Time Division–Synchronous Code Division Multiple Access
TE	Terminal Equipment
TIM	Traffic Indicator Map
TRAU	Transcoder and Rate Adapter Unit
TTA	telecommunications technology association
UDP	User Datagram Protocol
UE	User Equipment
UL	UpLink
UMTS	Universal Mobile Telecommunications System
UTRA	UMTS terrestrial radio access
UTRAN	UMTS terrestrial radio access network
VHE	Virtual Home Environment
VLR	Visitor Location Register
VMSC	Visited MSC
W–CDMA	Wideband Code Division Multiple Access
WEP	Wired Equivalent Privacy
Wi-Fi	Wireless Fidelity
WLAN	Wireless LAN

Appendix C

Figures list

Appendix D

Tables list