REPORT ITU-R F.2058

Design techniques applicable to broadband fixed wireless access systems conveying Internet protocol packets or asynchronous transfer mode cells

(2006)

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1 Introduction

There is an urgent need for broadband services in the telecommunication market. Users' needs are not uniform but diversified in terms of bit rate and service quality. Internet protocol (IP) and asynchronous transfer mode (ATM) are becoming basic signal transfer methods in recent wired networks in order to cope with such multimedia service demands. This trend also affects the fixed wireless systems used in the access networks. It is generally understood that IP-based and ATM-based fixed wireless access (FWA) systems should desirably have similar quality of service (QoS) or class of service (CoS) control capability to that of wired systems.

Both QoS and CoS are defined as classified categories prescribed by transmission performance parameters. QoS is used for classes of the transmission performance which satisfy users' quality requirements. On the other hand, CoS is generally used for the classes of similar performance which are presented to users as a guideline by a network administrator.

Furthermore, FWA systems are required to achieve objectives of IP or ATM performance parameters in the wired network adopted in ITU-T Recommendations I.356 and Y.1541. For this purpose, PHY layer parameters of FWA systems must be designed to meet these objectives of IP or ATM performance parameters.

Relations between PHY layer parameters and IP layer performance parameters, such as IPTD are given in ITU-T Recommendations Y.1540 and Y.1541. Relations between PHY layer parameters and ATM layer performance parameters, such as the cell loss ratio (CLR) or cell error ratio (CER) are given in ITU-T Recommendation I.356. ATM-based FWA systems should also satisfy the (SESR) objective specified in Recommendation ITU-R F.1668 (performance and availability requirements and objectives for FWA systems connected to PSTN are described in Recommendation ITU-R F.1400).

Both IP and ATM are packet or cell-based technologies operating at a high clock frequency where IP packet or ATM cell transmission may affect the system design and utilization efficiency of the radio-frequency spectrum. Such requirements will lead to design approaches which differ from existing voice-based FWA. Also necessary technologies for IP packets or ATM cells in FWA applications are, in many respects, common to those used in radio local area networks (RLANs).

The purpose of this Report is to provide example design techniques and methods addressing various aspects of FWA systems carrying IP or ATM data.

2 Types of FWA systems conveying IP packets or ATM cells in terms of radio channel utilization

Types of FWA systems conveying IP packets or ATM cells in terms of radio channel utilization can be categorized as follows.

The first type is to allocate an exclusive radio channel to each subscriber permanently or during a call retains. The typical system is point-to-point (P-P) or point-to-multipoint (P-MP) broadband wireless access systems generally using frequency bands above 20 GHz. This type of system is usually used for the leased line access services.

The second type is to allocate a radio resource when a call attempt occurs. The typical system is an FWA system providing PSTN services for mass users. Some of the systems use the technologies of mobile wireless access systems in order to reduce the cost of equipment.

The third type is to allocate a radio channel only when traffic such as IP packets or ATM cells occurs. This type achieves more efficient frequency usage compared with the first types especially for P-MP topology, provided that overhead bits are not too numerous. MP-MP topology is also suitable in this type of FWA.

The typical systems for the second type include those based on RLANs. FWA systems based on RLAN technologies may also have an advantage of reduce equipment costs. A feature of the third type may be explained as follows in case of ATM cell transmission. There are two cases for FWA systems conveying ATM cells:

a) FWA transmits ATM cells transparently without identifying any information contained in a cell;

b) FWA identifies information contained in the header portion of an ATM cell and thus controls the treatment of each cell within the radio section.

In case a), the system design method of FWA should basically be identical to systems intended for SDH signal transmission. This corresponds to the first type of FWA systems described above.

On the other hand, in case b), there can be a different approach in transmitting ATM cells over the radio section in order to efficiently utilize radio-frequency spectrum.

Several RLANs, used mostly for nomadic wireless access (NWA) purposes, or other types of wireless access systems using IP packet or ATM cell over the radio are realized as commercial products or examined in standardization bodies of forums. Standards for wireless access systems, both fixed and nomadic, are given in Recommendation ITU-R F.757 and Recommendation ITU-R M.1450.

3 Scope

This Report provides various design techniques applicable to FWA systems based on RLAN or NWA system technologies, which provide IP packets or ATM cell transport with a rate of more than several Mbit/s targeted for ubiquitous penetration.

The design solutions presented in the Annexes to this Report are suggested as one means towards satisfying the particular MAC layer standards set forth in ITU-T Recommendations concerning IP and ATM transmission. It should also be noted that the material presented in Annex 6 is not applicable to IP traffic since the idle cell notion is very specific to ATM.

The FWA discussed in this Report is focused primarily on P-MP applications wherein the capacity of one radio carrier provided by a base station will be shared by several or more terminal stations, although some of the techniques described in the Annexes to this Report are applicable also to P-P and MP-MP applications.

4 References

ITU-R Recommendations

Recommendation ITU-R F.757 Basic system requirements and performance objectives for fixed wireless access using mobile-derived technologies offering telephony and data communication services.

Recommendation ITU-R M.1450 Characteristics of broadband radio local area networks.

Recommendation ITU-R F.1668 Error performance objectives for real digital fixed wireless links used in 27 500 km hypothetical reference paths and connections.

Recommendation ITU-R F.1704 Characteristics of multipoint-to-multipoint fixed wireless systems with mesh network topology operating in frequency bands above about 17 GHz, 2004.

Recommendation ITU-R F.1763	Radio interface standards for broadband wireless access systems in the fixed service operating below 66 GHz.

ITU-T Recommendations

ITU-T Recommendation Y.1540	Internet protocol communication service – IP packet transfer and availability performance parameters.
ITU-T Recommendation Y.1541	Network performance objectives for IP-based services.
ITU-T Recommendation P.800	Methods for subjective determination of transmission quality.
ITU-T Recommendation P.862	Mapping function for transforming P.862 raw result scores to MOS-LQO.
ITU-T Recommendation I.356	B-ISDN ATM layer cell transfer performance.
ITU-T Recommendation G.107	The E-model, a computational model for use in transmission planning.
ITU-T Recommendation G.113	Transmission impairments due to speech processing.
ITU-T Recommendation G.114	One-way transmission time.
ITU-T Recommendation G.711	Pulse code modulation (PCM) of voice frequencies.
ITU-T Recommendation G.723.1	Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s.
ITU-T Recommendation G.729	Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP).

Others

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5 Abbreviations

This Report uses the following abbreviations:

ACK	Acknowledge
AIFS	Arbitration inter-frame space
ARQ	Automatic repeat request
ATM	Asynchronous transfer mode
BC	Broadcast channel
BCC	Block convolutional code
BE	Best effort
BEB	Binary exponential backoff
BER	Bit error ratio
BS	Base station
BWA	Broadband wireless access
CBR	Constant bit rate
CBQ	Class based queuing
CDV	Cell delay variation
CER	Cell error ratio
CLR	Cell loss ratio

CMR	Cell misinsertion ratio
CoS	Class of service
CSMA/CA	Carrier sensing multiple access with collision avoidance
CTD	Cell transfer delay
CW	Contention window
CWmin	Contention window minimum
DAMA	Demand assign multiple access
DiffServ	Differentiated service
DIFS	Distributed coordination function inter frame space
DL	Data link
DS	Differentiated service
DSA	Dynamic slot assignment
EDCF	Enhanced distributed coordination function
FCS	Frame check sequence
FDD	Frequency division duplex
FSDD	Frequency switched division duplexing
FEC	Forward error correction
FIFO	First in first out
FTP	File transfer protocol
FWA	Fixed wireless access
GBN	Go-Back-N method
GBR	Guaranteed bit rate
HCF	Hybrid coordination function
HEC	Header error control
HIPERACCESS	High performance radio ACCESS network
HIPERMAN	High performance radio metropolitan area network
HRP	Hypothetical reference path
IEEE	Institute of Electrical and Electronics Engineering
IETF	Internet engineering task force
IP	Internet protocol
IPDV	IP packet delay variation
IPER	IP packet error ratio
IPLR	IP packet loss ratio
IPTD	IP packet transfer delay
ITU-R	International Telecommunication Union – Radiocommunication Sector

ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
LCO	Loss cell outcome
LDP	Label distribution protocol
LSR	Label switching router
MAC	Media access control layer
MC	Multicast
MP	Measurement point
MP-MP	Multipoint-to-multipoint
MPLS	Multi-protocol label switching
MOS	Mean opinion score
NAK	Negative Acknowledge
nrtPS	Non-real-time polling service
OFDM	Orthogonal frequency division multiplexing
PCR	Peak cell rate
PDU	Protocol data units
PHSI	Payload header suppression identifier
PLC	Packet loss concealment
PS	Physical slot
P_{pl}	Packet loss probability
P_{plN}	Packet loss probability in network
P_{plB}	Packet loss probability in jitter absorption buffer
PHB	Per-hop behaviour
PHY	Physical layer
P-P	Point-to-point
P-MP	Point-to-multipoint
PESQ	Perceptual evaluation of speech quality
PSTN	Public switched telephone network
QoS	Quality of service
R	Rating factor
RSVP	Resource reservation protocol
rtPS	Real-time polling service
RTS/CTS	Request to send/clear to send
SLA	Service level agreement
SC	Single carrier
SDH	Synchronous digital hierarchy

SDMA	Space division multiple access
SIFS	Short inter frame space
SR	Selective repeat
TC	Traffic category
TCI	Tag control information
TDD	Time division duplex
TDM	Time division multiplex
TDMA	Time division multiple access
ToS	Type of service
TTC	Telecommunication Technology Committee
UBR	Unspecified bit rate
UGS	Unsolicited grant service
UNI	User network interface
VoIP	Voice over Internet protocol
WAN	Wide area network
WFQ	Weighted fair queuing
WRR	Weighted round robin

6 Technical characteristics addressed

6.1 Modulation and multiple access method

For guidance on preferred methods of multiple access and modulation techniques for broadband FWA systems conveying IP packets or ATM cells, Table 1 can be referred to.

TABLE 1

Methods of multiple access and modulation techniques

Conveyed signal	Multiple access(1)	Modulation
IP packet	TDM-TDD/FDD TDMA-TDD/FDD OFDMA-TDD/FDD DAMA-TDD/FDD CSMA/CA DSSS-OFDM	Adaptive (4-, 16-, and 64-QAM)(2) Adaptive (BPSK, 4-, 16-, and 64-QAM)(2) Adaptive (BPSK, 4- 16-, 64- and 256-QAM)(2) Presettable (4- and 16-QAM) CCK (QPSK) DSSS (BPSK, QPSK)
ATM cell	TDM-TDD/FDD TDMA-TDD/FDD OFDMA-TDD/FDD DAMA-TDD/FDD	Adaptive (4-, 16-, and 64-QAM)(2) Adaptive (BPSK, 4-, 16-, and 64-QAM)(2) Adaptive (BPSK, 4-, 16-, 64- and 256-QAM)(2) Presettable (4- and 16-QAM)

(1) Including duplex technique.

(2) These modulation techniques may be applied in conjunction with OFDM.

6.2 CoS and QoS

In the variety of broadband applications, there is greater demand for services with guaranteed service quality. It is also desirable to adopt service quality guarantee in FWA systems where communications qualities match the various services.

ATM-based FWA has the capability of QoS control which is one of capabilities of the ATM transport. Conversely, CoS control is a realistic solution in the IP-based FWA systems because of the limited bandwidth and characteristics of the radio channel.

Annex 1 describes a mechanism to provide CoS and QoS control in FWA systems using distributed control or centralized control.

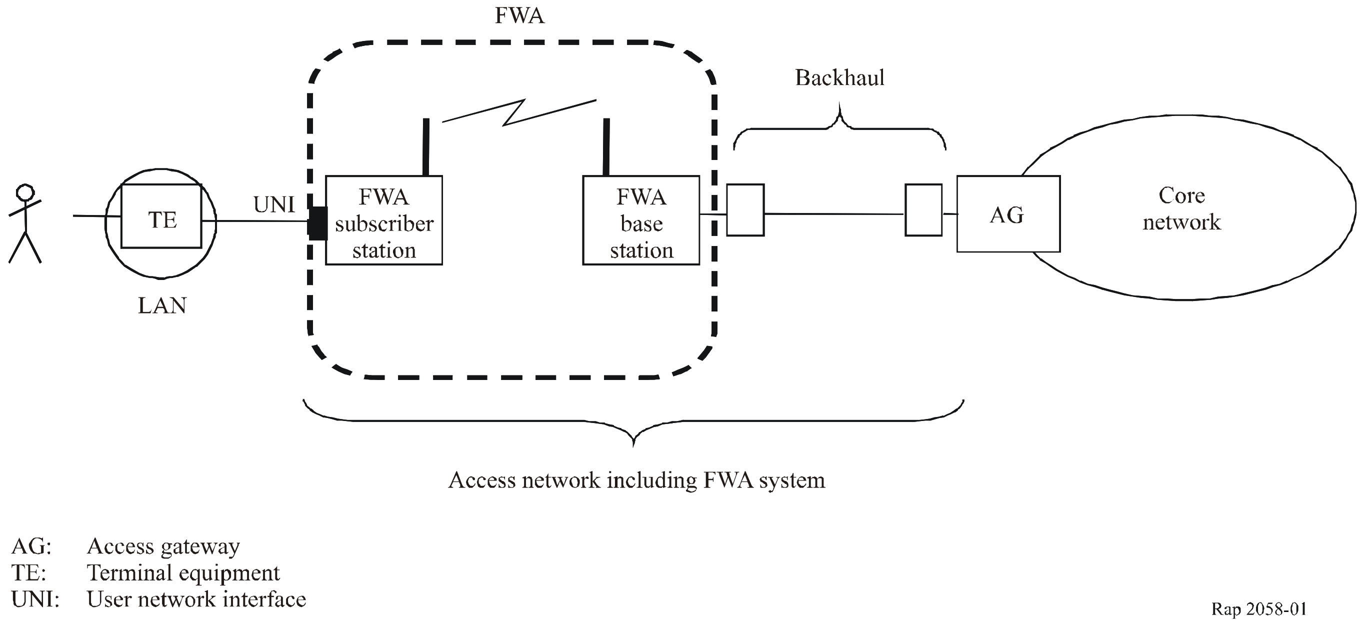
6.3 Transfer delay characteristics

ITU-T Recommendation Y.1541 specifies overall IP performance objectives. Among the performance parameters, transfer delay parameters, i.e. IPTD and IPDV defined in ITU-T Recommendation Y.1540, should be taken into account in the design of FWA systems conveying IP packets or ATM cells.

For the time being, there is no specification on how the UNI end-to-UNI end requirements for IPTD and IPDV are sub-divided into individual portions forming the access network. Basically, FWA systems should be designed so that the overall IPTD and IPDV could meet, e.g. in the configuration in Fig. 1, the requirements specified in ITU-T Recommendation Y.1541.

FIGURE 1

Use of an FWA system in the access network



In order to achieve the above requirements, the following considerations are noted.

In general, FWA equipment (e.g. subscriber station, base station, and so on) may have some buffers for controlling transmission timings, compensating jitters, formatting frame, and so on. Such buffers lead to delay degradation in FWA systems. Representative access protocols for FWA systems are CSMA/CA and TDMA, and such access protocols may become the dominant factors of delay characteristics in FWA systems.

6.3.1 Considerations on IPTD

In CSMA/CA scheme, for packet collision avoidance, backoff and RTS/CTS sending techniques are adopted. Those techniques cause IPTD degradation, especially when many stations try to send packets simultaneously (see Annex 2).

In TDMA scheme, base stations control the entire bandwidths and assign them to each service flow, so that collision-free connections can be realized in principle. Once an initial connection procedure can be finished, IPTD degradation like CSMA/CA due to collision avoidance like backoff and RTS/CTS may not happen. However, several factors like scheduling algorism, frame structure, traffic loads and capacity may become causes of IPTD degradation (see Annex 3).

6.3.2 Considerations on IPDV

CSMA/CA-based FWA systems usually adopts BEB scheme. It may cause large IPDV occasionally. This scheme mitigates probability of simultaneous re-transmissions from multiple stations, but incurs IPDV degradation. In addition, especially in case that a long packet occupies the radio channel for a long span, waiting time which happen to other packets to be transmitted will increase. This also leads to IPDV degradation.

In TDMA scheme, base stations can control bandwidths on service flow basis. Thus, low-jitter transmission can be realized compared with CSMA/CA.

For technical guidance and examples of calculation on IPTD and IPDV occurring wireless system to access portion of the IP network (see Annex 4).

6.4 VoIP technique

VoIP is recently regarded as one of the most attractive solutions of IP-based network.

There are several performance parameters for providing a certain QoS for VoIP. Some of the parameters, such as delay or packet loss ratio, are defined in the ITU-T Recommendations for the end-to-end network (including both wired and wireless sections). Objectives for such performance parameters in wireless sections should also be defined.

Annex 5 provides technical characteristics of broadband FWA systems to support VoIP, as derived from studies of the TTC that can be found in TTC Standard JJ-201.01 – A Method for Speech Quality Assessment of IP Telephony, 2003.

6.5 Efficient spectrum utilization technique

Two possible control methods to utilize the spectrum efficiently consist of PHY layer control technique and MAC layer control technique.

The PHY layer control technique is further classified in two ways. One way is to fill packets or cells full by applying adaptive modulation, adaptive TDD and DAMA. The other way is to increase the efficiency of spectrum utilization by applying SDMA using sector antenna or array antenna.

The MAC layer control technique assigns adequate spectrum to data according to its application or service quality. Data is fragmented or packed to increase the efficiency of data load utility. Annex 6 describes a technology to remove idle cells without introducing CDV for ATM-based FWA to achieve efficient spectrum utilization.

6.6 Error correction techniques in an ATM-based FWA system

In ATM-based FWA systems, ARQ and/or FEC effectively improve error performance.

Annex 7 provides the error correction techniques in an ATM-based FWA system.

Annex 1

Mechanism to ensure QoS or CoS in the broadband FWA systems

1 Introduction

In this Annex, mechanisms to ensure QoS or/and CoS in the broadband FWA systems are described. There are two types of radio channel access schemes. One is the distributed control system and the other is the centralized control system. An example of the distributed control system is the IEEE 802.11 systems. They are packet switching service adopting CSMA/CA. On the other hand, a base station assigns the time slot of the TDMA MAC frame to subscriber terminals or a combined frequency and time partition in OFDMA in the centralized control system. Examples of the centralized control system are ETSI-BRAN HIPERMAN, HIPERACCESS, HIPERLAN, IEEE802.16-2004 or MMAC-HSWA HiSWAN. Full QoS, even in overloading conditions, is achievable only with centralized control systems.

There are two schemes to achieve QoS and CoS. One is the prioritized scheme which offers a priority control among the service classes without specifying service specific parameters. The other is a parameterized scheme to assure required communication quality parameters. Only a parameterized scheme has the possibility to guarantee QoS.

First of all, an overview of CoS control is presented in § 2. Subsequently, queuing and priority management mechanisms to support CoS controls are shown in § 3. Then, a CoS control mechanism for a distributed control scheme and a QoS control mechanism for a centralized control scheme are presented in § 4 and 5, respectively. Finally, a comparison of distributed control scheme system and a centralized control scheme system is shown in § 6.

2 Overview of CoS control

2.1 Airplane model

CoS control is often explained by an "airplane model" (see Fig. 2). Service quality is classified into several service classes just like airplane seats which are classified into first, business and economy class. Higher service classes than usual best effort class are used to offer high-level services, e.g. ensuring minimum delay time or available bandwidth. High-quality service is provided if the request from the user is accepted. Admission control or policy control methods are used to determine which service classes are allowed for data transfer. According to the service class, each data transfer is transferred based on that quality. However the amount of traffic carried in such higher service classes is limited because the available bandwidth is limited.

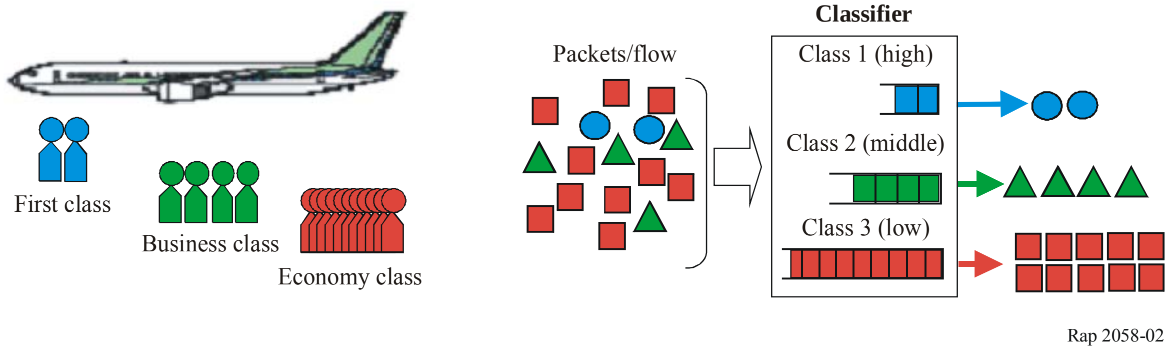
2.2 CoS control mechanism for the best effort flows in the FWA environment

2.2.1 Priority control for the CoS control

At least two CoS are necessary to offer CoS control. Priority is associated to the CoS and service differentiation is made according to the priority. The base station (BS) must determine the CoS for each best effort transfer. A subscriber needs to know the CoS in transmitting its best effort data. The methods by which CoS is determined and translated to the subscriber are introduced in the following paragraphs.

FIGURE 2

Airplane model



2.2.2 Priority determination

There are several ways to determine the CoS for transmitting best effort data between the base station and a subscriber station. The business users or high end customers may have higher priority in transmitting the best effort data than others. In this case, an authentication mechanism is used to determine the CoS of the subscriber. The base station may control the bandwidth allocated to a subscriber that occupies the system resources by transmitting or receiving a greater amount of data. To protect the systems resources from greedy subscribers, the base station may change the CoS for such a subscriber to low. In this case, the base station checks the amount of data transmitted and/or received by each subscriber and changes the CoS based on the amount of data and determined threshold.

In both cases, it is the role of the base station to determine the CoS of a subscriber. In the former case, the base station can ask the authentication server or may check the internal database. In the latter case, the base station can control the CoS offered for each subscriber.

2.2.3 Notification of the CoS to subscribers

To make CoS control in both forward and reverse directions, the base station needs to note the CoS of each subscriber. For this purpose, outbound or inbound signalling is used.

In the outbound signalling scheme, the base station sends the service class information to each subscriber before data transfer occurs. In this case, the base station must also notify a subscriber of any changes to its CoS.

Another alternative is the inbound signalling scheme where the base station sends the determined information on the CoS together with the data by using a similar mechanism defined in like IEEE 802.1D Annex H2 standard as shown in Fig. 3. The base station sets the priority of a frame in the frame header. Since the priority is associated with the CoS, the subscriber can recognize the CoS from the priority information.

As in Fig. 4, the subscriber sets the same priority information with the latest received data frame.

The determined CoS must be considered in the access control method. Although there are many random access methods, most of them do not support such functionality. A CSMA/CA-based method has been proposed to offer prioritized transmission.

A priority control scheme by the inbound signalling scheme seems to be a realistic solution for CoS control in the FWA systems as it does not need too much bandwidth compared to the outbound signalling scheme.

FIGURE 3

Priority control with frame tagging

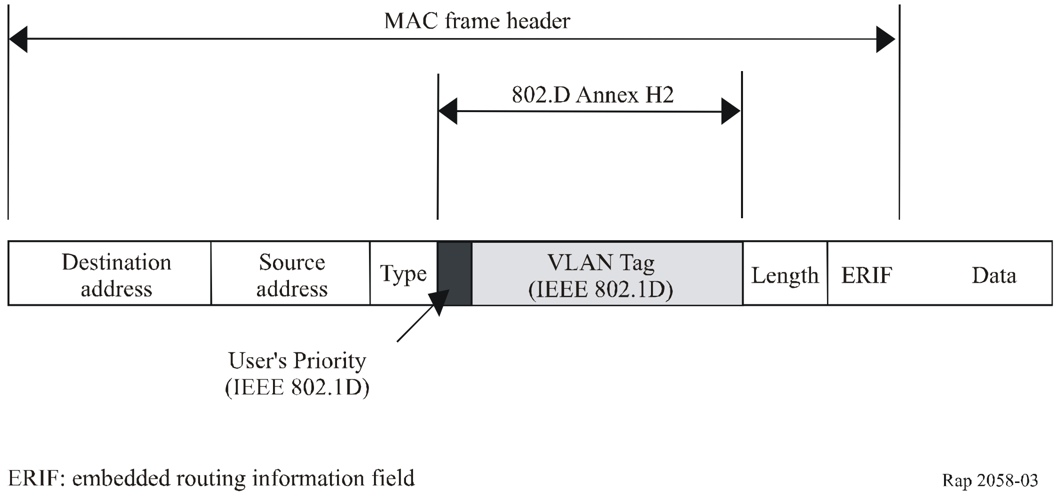
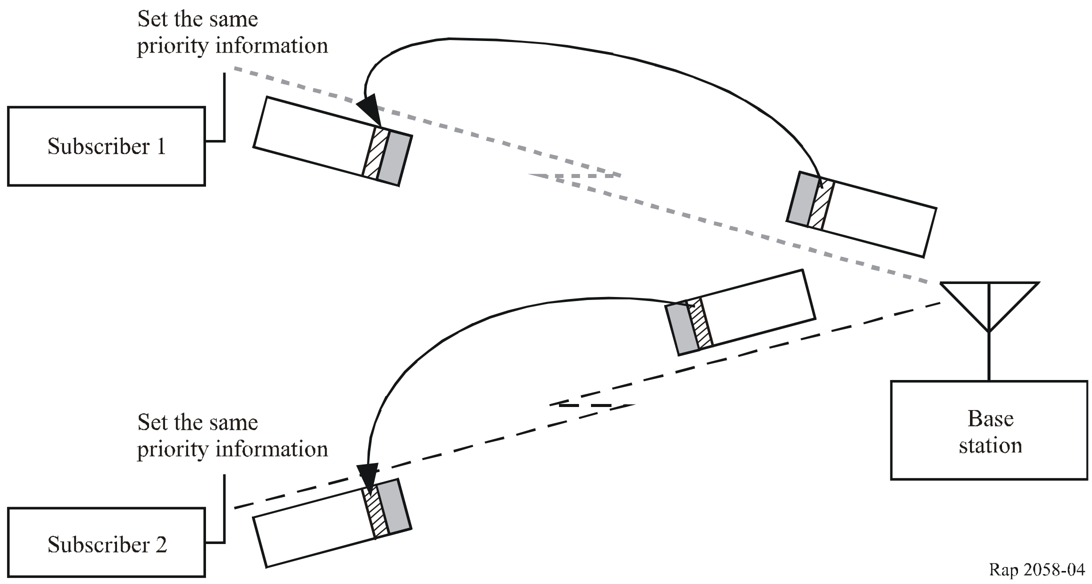


FIGURE 4

CoS notification by the inbound signalling scheme



2.3 Effect of CoS control

2.3.1 Fair access

One useful way of CoS control is to prevent a specific user from occupying the limited bandwidth of the wireless channel and to provide fair access for all subscribers. By making a lower service class than the usual best effort class, a subscriber who consumes most of the frequency resource alone will be forced to degrade the service class.

Checking the amount of transferred data for each subscriber at the base stations, the operator can know the activity of each user. There will be some ways to check the amount of transferred data for each subscriber. For example, IP address may be enough to distinguish the heavy user. Of course, the operator can check the traffic flow in more detail.

The service provider must have a policy control mechanism to protect the system resources from a user who intentionally overly occupies the system resources.

2.3.2 End-to-end CoS control

To offer CoS controls in the end-to-end region, FWA base stations and/or their upper routers need to have CoS handling functions and, if possible, subscriber terminals should also have a corresponding functionality. Whether CoS control is accomplished or not depends on the available bandwidth. To meet the various users' requirements of subscribers, the system needs much greater bandwidth than necessary for usual best effort services. Although the bandwidth limited and the radio channel is not so reliable, certain level of CoS control will nevertheless be offered.

3 Queuing and priority management mechanisms to support CoS controls in the FWA systems

3.1 Introduction

This paragraph introduces the queuing mechanisms and the priority management scheme that support CoS controls in the FWA systems. A queuing mechanism that offers priority control and/or fair data transmissions will be used to make service differentiation according to service classes. Some kinds of priority queuing mechanisms are briefly described. Moreover, the priority management scheme is proposed to make fair data transfer by using the CoS control mechanism.

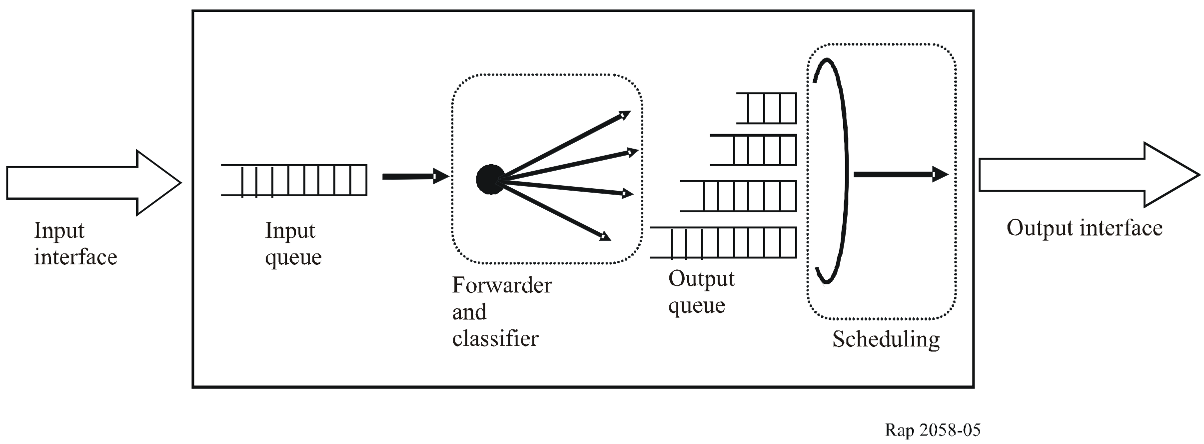
3.2 Queuing mechanisms for FWA systems

CoS control is the mechanism to ensure the communications quality of each data transfer according to its service class. CoS control is achieved when all nodes between the end systems have the function to make service differentiation according to the service class. The basic mechanism for this purpose is the queuing.

Queuing can be seen as a buffering operation to reorder the IP packets and plays a very important role in the forwarding nodes. This operation is carried out when a node relays a packet from the input queue of an interface to the output queue of another interface as in Fig. 5. In this process, the packet classification and the packet scheduling of the output queue plays a very important role in supporting CoS control. In this paragraph, the output queuing methods, i.e. the packet scheduling methods, are considered and some basic mechanisms are briefly introduced.

FIGURE 5

Queuing and packet scheduling in a forwarding node



3.2.1 First in First out (FIFO) queuing

FIFO queuing is the standard method in carrying the IP packet from input interface to the output interface in a store and forward manner. In FIFO queuing, incoming packets are put into the queue in the order of arrival and the packets are sent to the output interface in the same order.

FIFO queuing is the most popular mechanism and many vendors implement it in their products. Although the FIFO queuing is simple and fast mechanism, it suffers from queuing delay when the traffic load increases. Moreover, it cannot make service differentiation and therefore has difficulty to support CoS control functions.

3.2.2 Priority queuing

In the priority queuing, high-priority packets are always handled before the others. Recognizing the traffic type and putting the high-priority packet to the head of output queue achieve priority queuing. Several output queues corresponding to the offered traffic classes may be used. In the priority queuing, arrived packets are put into the output queue according to the determined order. Therefore, the higher-priority packets are transmitted before the lower priority ones.

3.2.3 Class based queuing (CBQ)

In the CBQ, traffic in the network is classified into some classes which are defined by the network operator to make different forwarding operations according to the traffic types. A packet is forwarded according to its traffic class. The forwarding nodes have output queues for each traffic class and the operator can schedule the outgoing traffic for each queue. On the contrary to priority queuing, CBQ gives every packet a chance to be forwarded according to the classified level of transmission.

3.2.4 Weighted fair queuing (WFQ)

WFQ is a combined mechanism of priority queuing and fair queuing. It realizes fair data transfer considering the priority and the amount of traffic for each traffic class.

3.3 CoS control schemes on layer 3 or higher

To support CoS control between end users, some mechanisms have been proposed which work on the IP layer. These schemes are effective to provide end-to-end CoS control. If the FWA base stations or the routers in the system support the functionality, various multimedia services can be provided. Current CoS control methods for layer 2 and 3 or higher are shown in Table 2.

TABLE 2

Current/CoS control methods

Layer	Method		Standardization
Layer 2	IEEE 802.1D Annex H2	IEEE 802.1D Annex H2 Tagging	IEEE 802 Committee
Layer 3-4	MPLS	Label Switching	IETF
	DiffServ	Use of DS field (IP ToS field)	IETF
	RSVP	RSVP signalling	IETF

3.3.1 Multi-protocol label switching (MPLS)

The label switching technique is used to make high-speed packet forwarding. A label which can be referred from the data link layer is assigned to the data flow to distinguish from the others. LSRs are used in MPLS in which packet forwarding is carried out not by the layer 3 but by the layer 2 checking the label in the frame header of the packets. Therefore, high-speed packet switching is enabled because IP header analysis is not necessary except for the first packet.

When assigning a label to a data transfer, service quality for that transfer can also be assigned. The label for a data transfer is propagated to other LSRs by the LDP. At this time, service class is also distributed, and CoS is provided by the LSRs. To support end-to-end CoS control, an edge router, the LSR which connects a conventional router and other LSRs must also support DiffServ model.

3.3.2 Differentiated services (DiffServ)

In the DiffServ model, service levels are distinguished by ToS field in the IP packet header. ToS field is not used in IPv4 and the meaning of that field is redefined in the DiffServ model. Policy or rule based traffic control is offered in the DS domain, which is constructed by the DS capable nodes. The operations of DS nodes such as packet forwarding manner for each data flow of the specific CoS is defined by the PHB.

When transferring data over DS domains with specific service quality, policy servers of those DS domains negotiate with each other. Necessary bandwidth for the data transfer is acquired by a bandwidth broker if the SLA is accomplished.

3.3.3 Policy-based network management for CoS provision

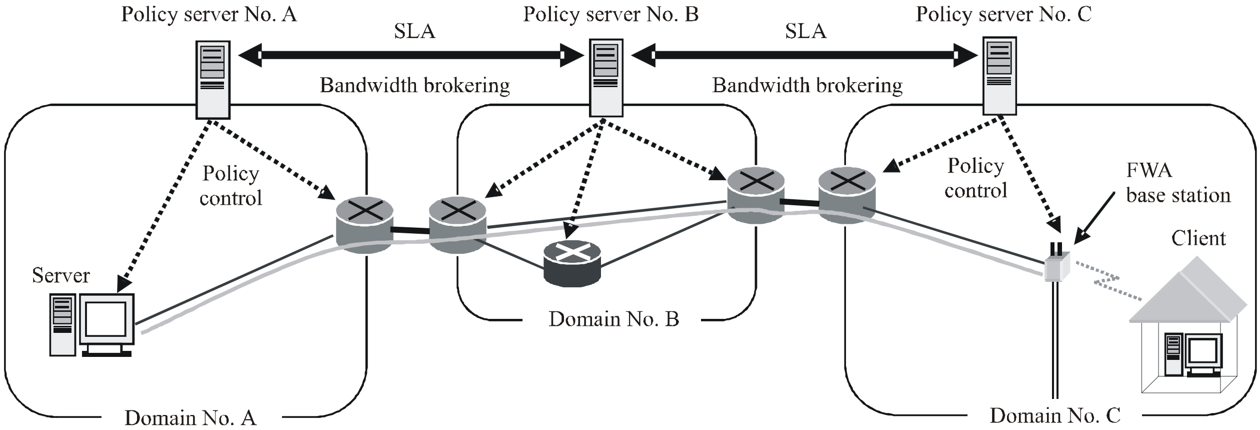
To determine a priority of a data transfer according to the requirement of subscriber or to control a subscriber's priority, policy or rule-based network management is necessary. The policy controller and admission controller will be used to manage the network resources or the subscribers' priority.

When policy-based network management is employed, centralized access control mechanism is desired at the layer 2 protocol in FWA systems. Although random access, such as CSMA or variations of it, may be enough to provide best effort services, centralized access mechanisms, such a polling based protocol, will be necessary to propagate and ensure the determined CoS for the data flow.

An example is illustrated in Fig. 6.

FIGURE 6

Policy-based network



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3.3.4 CoS control mechanism in the FWA systems

Some strategies can be considered to implement the CoS control mechanism in the FWA systems according to the functionality of the FWA base station. Basically, the MAC layer will have CoS control functionality for a data flow in the wireless region because the access procedure is defined there.

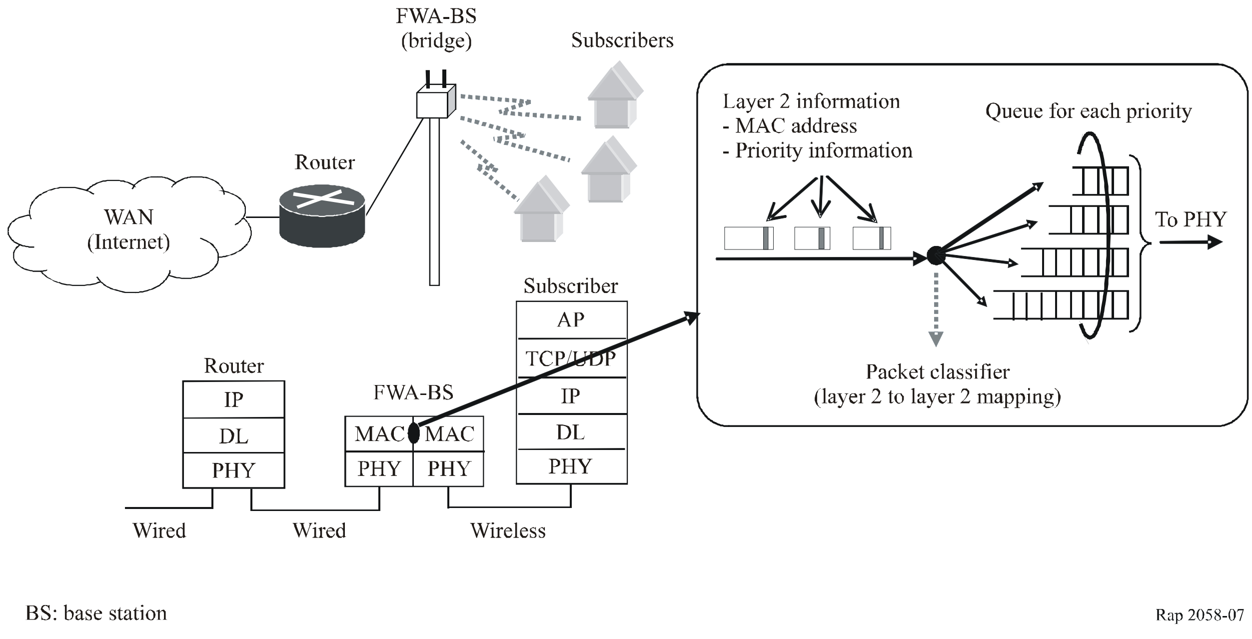
3.3.4.1 When an FWA base station is a bridge

When an FWA base station works as a bridge as shown in Fig. 7, a straightforward way to realize CoS is to have some queue corresponding to each service class. From the priority information in the MAC frame header, the base station judges the priority of the frame and queues it appropriately. If the base station relays the frame from the wired region to wireless, or vice-versa, protocol conversion may be required and the above operation will be accomplished in that process. When transmitting the buffered frame, the base station takes a frame from the buffer according to the determined algorithm and then sends it to the channel. One method to ensure CoS may be the queuing and priority management methods.

Other approaches to realize CoS is for further study.

FIGURE 7

FWA base station (BS) with bridge function

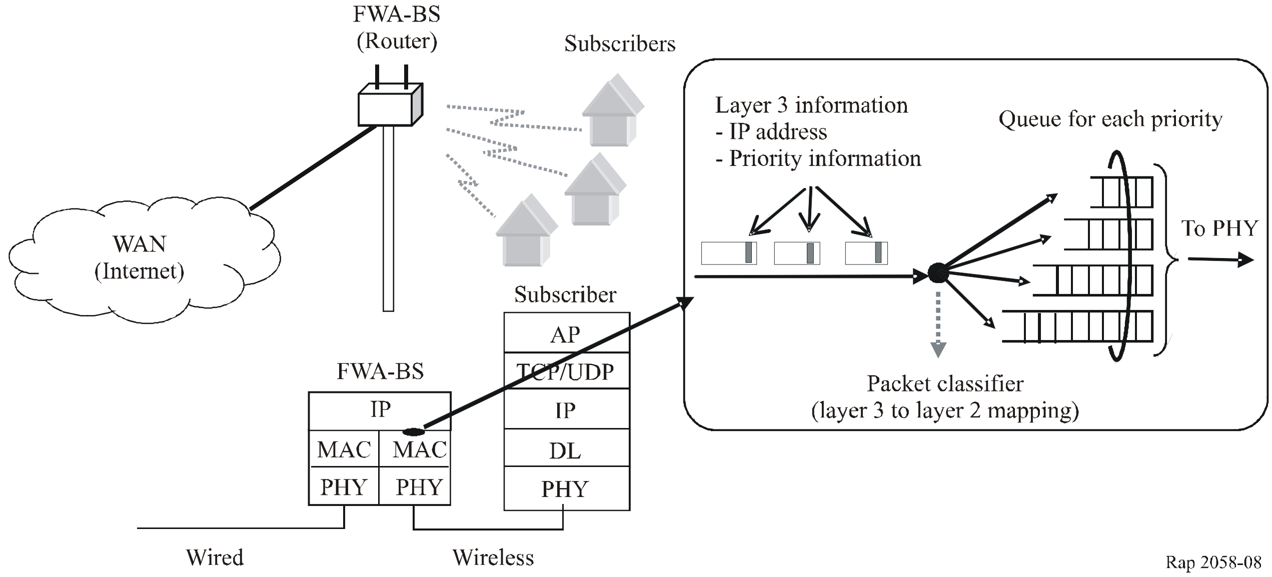


3.3.4.2 When an FWA base station is a router

When an FWA base station works as a router, an example to provide CoS control is described in Fig. 8. In this case, the base station must handle the priority information between the different layers, i.e. layer 3 and layer 2. In many schemes, an inbound signalling method is employed to distinguish the high priority packet from the usual best effort packets. In this case, the priority information is written in the packet header. An FWA base station is assumed to have priority queuing functionality in the MAC layer and mapping of the service class in the layer 3 and priority in the layer 2 is accomplished.

FIGURE 8

FWA BS with router function



3.3.4.3 Subscriber terminals

A subscriber terminal of the FWA system will require some mechanisms for CoS controls.

When receiving a data frame, the MAC layer of the subscriber terminal must understand the priority information in the frame and pass it to the upper layer. If a subscriber wants to send high priority data, the MAC layer of the terminal must set the priority information of the frame.

When a subscriber terminal is the source of high priority data, it may be required to have the functionality to negotiate with the policy server, the router or the base station to reserve the bandwidth for the data.

4 CoS control mechanism of FWA systems using a distributed control scheme

4.1 Extension of CSMA/CA protocol to support CoS controls

CSMA-based methods are widely used as distributed MAC mechanism in the LAN environment. The IEEE 802.11 wireless LAN system employs CSMA/CA protocol as its basic access method. CSMA/CA protocol provides equal opportunity to transmit data for the stations and priority of data is not considered. To support CoS controls, CSMA/CA protocol must be modified to make service differentiation considering the priority of transmitting data. Data priority is established by changing the backoff procedure of CSMA/CA protocol. The original and modified CSMA/CA protocol is briefly explained in this paragraph.

4.2 Original CSMA/CA protocol

In the CSMA/CA protocol, both the base station and subscribers continuously sense the channel to know whether the channel is available. The channel is deemed idle if the station does not detect a signal for a specified time interval called DIFS.

When data arrives at a station, it starts to transmit the data immediately if the channel is sensed idle. If the channel is busy at the time of arrival, the station carries out a backoff procedure once the channel becomes idle in order to avoid collisions. In the backoff procedure, a station that is ready to transmit data generates a random backoff time before transmission by generating a random number

N from a determined range. The station then begins to decrease the backoff time. The random number is decreased in every period called "slot time". The station initiates transmission of the frame if its backoff time becomes 0. In the case that the channel became busy again before the number reaches to zero, the station stops decreasing the backoff time and waits for the channel to be idle. When the channel does become idle again, the station once again starts to decrease the balance of its backoff time.

A station that received a data frame correctly returns an ACK to the sender in a specified time period called SIFS. If the sender does not receive an ACK within a specified time, it carries out retransmission of previous data frame.

4.3 Modified CSMA/CA for service differentiation

In the modified CSMA/CA protocol, a base station is assumed to have at least two queues for the best effort traffic. Each queue has its own priority to transmit data and the priority corresponds to the service class. The algorithm of the modified CSMA/CA is described assuming two service classes for the best effort transmissions. The service class that has higher priority for data transmission is called high priority class and the service class with low priority is called low priority class.

To make service differentiation, the backoff algorithm of CSMA/CA protocol is modified as shown in Fig. 9. In the modified CSMA/CA protocol, the stations use different slot times according to the service class. In Fig. 9, data exchange is carried out in high priority class between the base station and the subscriber 1 and the value of the slot time these stations use in the backoff algorithm is set T_A. Similarly, the base station and subscriber 2 make data exchanges between them in low priority class and set the value of slot time T_B. As in Fig. 9, T_A is smaller than T_B. By using different slot time values, high priority data tends to be transmitted more frequently than the low priority data. Therefore, the priority of the service class is reflected in the number of transmission attempts per service class.

The base station has two transmission queues corresponding to the service class of the data. It carries out the CSMA/CA procedure independently for each transmission queue. A subscriber station transmits data in their service class specified by the base station. As a result, service differentiation according to service class is achieved.

4.4 CoS control based on EDCF in an FWA system

4.4.1 Legacy DCF

The basic channel access procedure of the IEEE 802.11 wireless systems is a DCF (Distributed Coordination Function) known as CSMA/CA shown in Fig. 10. The CSMA/CA protocol provides equal opportunity to transmit data for the stations that are ready to do so and priority of data is not considered. In the CSMA/CA protocol, frames are transmitted in a distributed manner.

A station about to send data senses the channel before transmitting. The channel is considered to be idle if no carrier is detected for more than the carrier sense time DIFS. A station starts transmission immediately if the channel is idle. If the channel is not, the station performs the backoff procedure once the channel becomes idle and generates a random number for backoff timer. If the channel continues to be idle, the station decrements the backoff timer at specified intervals, called "Slot Time", within the CW, and transmits the data when the backoff timer reaches 0.

A station that successfully receives a frame returns an ACK to the sender SIFS time after the reception. The sender of data frames retransmits the frame if the ACK is not returned within a specified time.

FIGURE 9

Service differentiation by using CSMA/CA with prioritized transmission

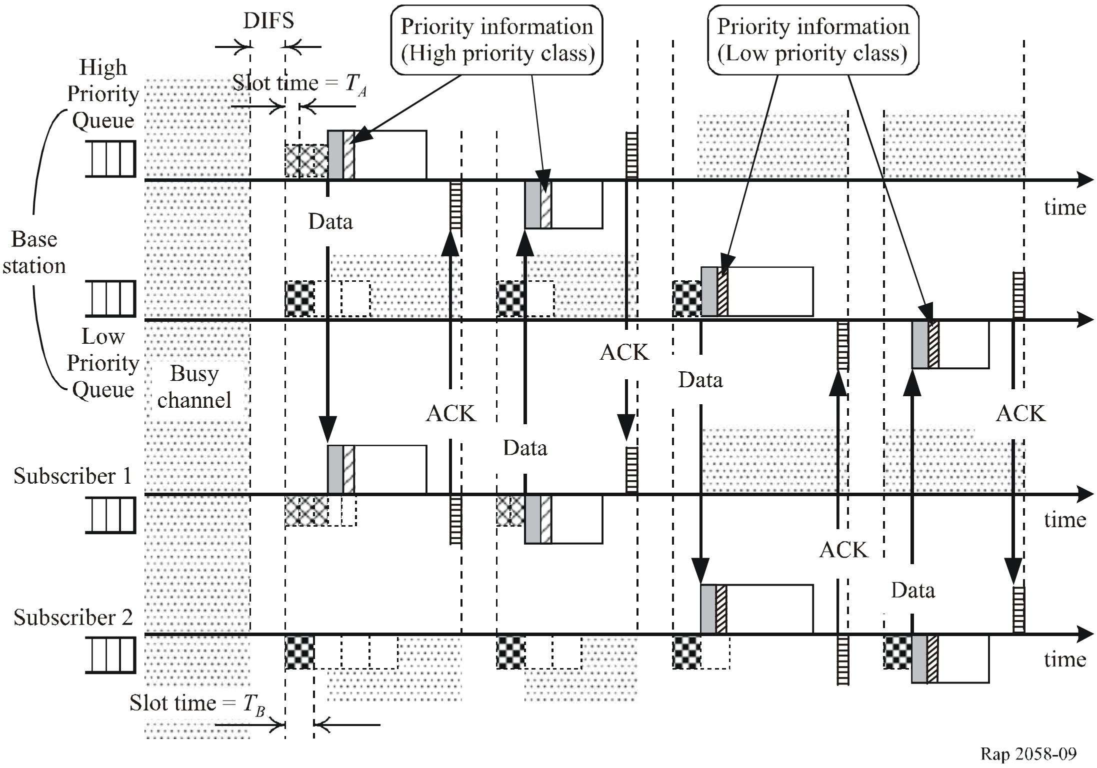
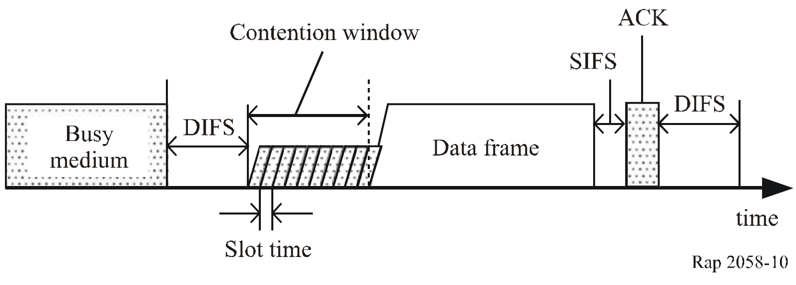


FIGURE 10

An example of IEEE 802.11 channel

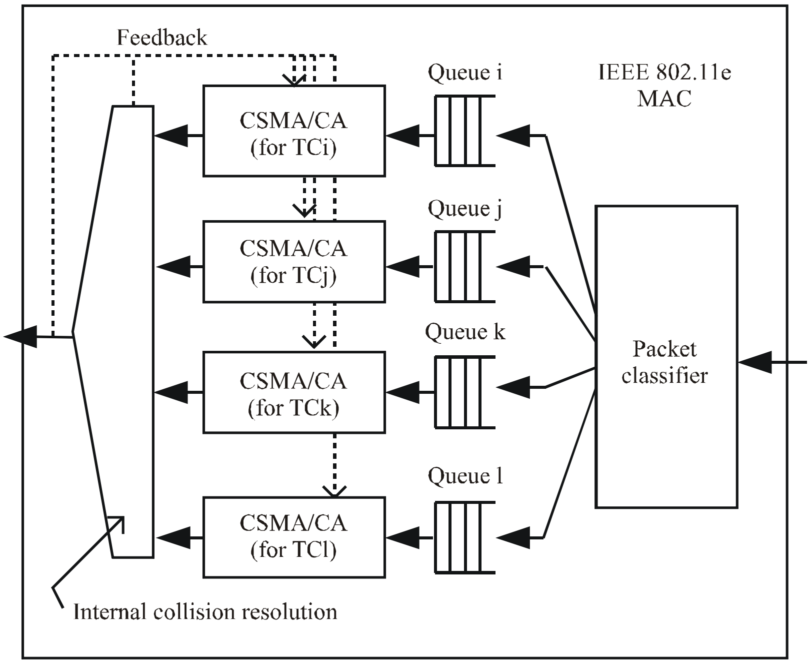


4.4.2 Enhanced distributed coordination function (EDCF)

EDCF is the modified CSMA/CA mechanism. In EDCF, data frames are classified into at most eight TCs where the number of TCs match up with those defined by IEEE 802.1D Annex H. Figure 11 shows the structure of IEEE 802.11TGe MAC. EDCF stations have at most eight prioritized output queues, one for each TC. When an EDCF station starts to send the data frame, some output queues compete for opportunity to transmit data frame each other by using EDCF procedure. The EDCF protocol provides differentiated CSMA/CA access to the wireless medium for prioritized output queues and makes service differentiation considering the priority of data frame. In EDCF, this service differentiation is realized by using two priority control methods described below.

FIGURE 11

IEEE 802.11e MAC



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4.4.3 Backoff algorithm

The backoff procedure is based on the binary backoff algorithm. Backoff time equals the slot time multiplied by a random number. In the backoff procedure, station generates a random number from a uniform distribution ranging from 0 to CW. Every time station retransmits the frame, CW value is sequentially ascending integer powers of 2, minus 1, until the CW reaches the value of CWmax. Once the CW reaches CWmax, it remains at the value of CWmax until it is reset.

In EDCF procedure, a station calculates and maintains CW for each prioritized queue, i.e. queue i: 0~CWi, queue j: 0~CWj. If priority of queue i is higher than that of queue j, CWi is set to smaller than CWj. By using this procedure, service differentiation is realized.

4.4.4 Arbitration inter-frame space (AIFS)

In legacy CSMA/CA procedure, the channel is considered to be idle if no carrier is detected for more than the DIFS. EDCF procedure uses AIFS instead of DIFS as shown by Fig. 12. If priority of queue i is higher than that of queue j, AIFS[i] is shorter than AIFS[j]. By using this procedure, priority control for some TCs is realized.

4.5 An example of CoS service class (IEEE 802.1D Annex H2)

In Ethernet based systems, eight levels of service qualities are considered in IEEE 802.1D Annex H2 and each service quality is mapped to a user priority. The information on the user priority is contained in the extended header field of a MAC frame. In this scheme, both prioritized CoS and parameterized CoS are supported. As in Table 3, user priorities of 4, 5 and 6 are parameterized CoS and the others are prioritized CoS.

FIGURE 12

Priority control mechanism by using AIFS

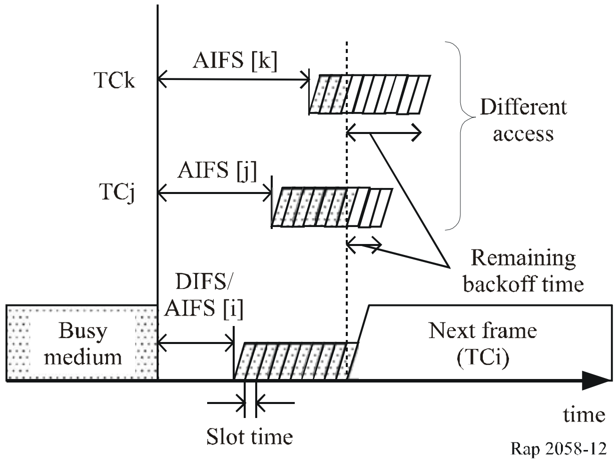


TABLE 3

User priority and traffic categories in the IEEE 802 LANs

User priority	Traffic type	Description
1	Background (BK)	Bulk transfers and other activities that are permitted on the network but that should not impact the use of network by other users and applications
2	Spare	
0 (default)	Best effort (BE)	LAN traffic, as we know today
3	Excellent effort (EE)	The best effort type services that an information services organization would deliver to its most important customers
4	Controlled load (CL)	Important business applications subject to some form of "admission control" are that pre-planning of the network requirement at one extreme to bandwidth reservation per flow at the time the flow is started at the other
5	Video (VI)	Less than 100 ms delay
6	Voice (VO)	Less than 10 ms delay, and hence maximum jitter. (One way transmission)
7	Network control (NC)	To maintain and support the network infrastructure

Many products in the market support this mechanism by using a priority queuing mechanism such as weighted fair queuing (WFQ) and weighted round robin (WRR).

The IEEE 802.1D Annex H2 refers to a mapping mechanism between the traffic types and the traffic classes depending on the number of queues a device has. Table 4 shows the mapping scheme in that standard.

TABLE 4

Traffic type to traffic class mapping

Number of queues	Traffic types
1	{BK, BE, EE, CL, VI, VO, NC}
2	{BK, BE, EE}, {CL, VI, VO, NC}
3	{BK, BE, EE}, {CL, VI}, {VO, NC}
4	{BK}, {BE, EE}, {CL, VI}, {VO, NC}
5	{BK}, {BE, EE}, {CL}, {VI}, {VO, NC}
6	{BK}, {BE}, {EE}, {CL}, {VI}, {VO, NC}
7	{BK}, {BE}, {EE}, {CL}, {VI}, {VO}, {NC}

5 A mechanism for FWA systems using a centralized control scheme

5.1 Introduction

Recently, QoS control has been a key technology to support multimedia traffic in IP networks. With QoS control, a system can provide various levels of communication qualities depending on the requirement of the user. The system needs complex mechanisms such as bandwidth reservation/allocation, admission control, policy control and bandwidth brokering to achieve QoS.

The centralized system is suitable for achieving QoS. It has the scheduling function for bandwidth assignment in a base station.

The bandwidth is assigned according to applications such as telephone, e-mail, video, etc. Since their bandwidth and delay requirements are different, the base station prepares three classes of QoS. Paragraph 5.2 presents an example of a specification of QoS class in wireless systems. Paragraph 5.3 then presents a scheduling function for a bandwidth assignment in the base station.

5.2 An example of QoS class in a wireless system

An example of QoS class in a wireless system using a centralized control scheme is presented in Table 5.

TABLE 5

An example of QoS class of wireless systems using a centralized control scheme

QoS Class	CBR/GBR/UBR	ARQ	Example of service
1	GBR or UBR	With	E-mail, FTP
2	CBR	With	Image
3	CBR	No	Telephone, real-time image

Class 1 provides guaranteed bit rate (GBR) or unspecified bit rate (UBR) service which guarantees the minimum bandwidth assigned to a subscriber station while keeping best-effort type communication. This class assures the availability of a specified minimum bandwidth even though the traffic from many users may be combined.

Classes 2 and 3 provide CBR service as per the user's declaration. They are suitable for receiving real-time imaging services. Class 3 does not use ARQ in an effort to minimize the data transfer delay.

5.3 Calculation of effective bandwidth for different services

Since different services vary in their natures, corresponding to different bandwidth requirement, the system should be available to distinguish different service classes and should decide methods of bandwidth assignment for each service. Such a method to achieve this is illustrated below in detail. Firstly, services should be sorted into several kinds according to their attributes, such as delay requirement or priority requirement. Secondly, for each kind service, one corresponding equation would be applied to calculate its effective bandwidth, implying the QoS requirement of the services. Lastly, the system would assign the resulted bandwidth to each kind service. In this way, the QoS of services could be achieved in a centralized FWA system.

One typical characteristic of the packet data traffic is burstiness, and the effective bandwidth is defined here for characterizing the general amount of resources utilized by packet data traffic. In addition, it is also necessary to specify average bandwidth and maximum bandwidth. As the data of a certain kind of service is being transferred, the system should dynamically adjust the bandwidth for transmitting, and then evaluate the transmitting quality by obtaining such parameters as delay, BER, etc. Also the user's subjective judgement should be one of criteria for obtaining average bandwidth. If the received data is just acceptable, the required bandwidth at this time equals to the average bandwidth. If no significant performance improvement could be achieved along with bandwidth increase, the corresponding bandwidth is the maximum bandwidth.

The bursty nature of packet data may lead to possible losses during temporary overloads. Therefore the base station should reserve as more resources as possible to achieve the best performance. On the other hand, operators should desire efficient usage of available resources and should avoid allocating too much width to one user. To utilize resource more efficiently, one method for calculation of effective bandwidth has been proposed in the following content and four kinds of services are illustrated.

Conversational service

The basic qualities of this class are low delay, low jitter (delay-variation), reasonable clarity, and absence of echo. In the case of multimedia applications, it is also necessary to maintain relative timing of the different media streams.

As one CBR service, its max throughput is the same as average throughput and its effective bandwidth should be calculated as the following equation:

$$Bandwidth_{effective} = Th_{average} + \varphi * (TH_{max} - TH_{average}) \qquad (1)$$

where:

$Bandwidth_{effective}$: effective bandwidth of service

$Th_{average}$: average bandwidth of service

TH_{max}: max bandwidth of service

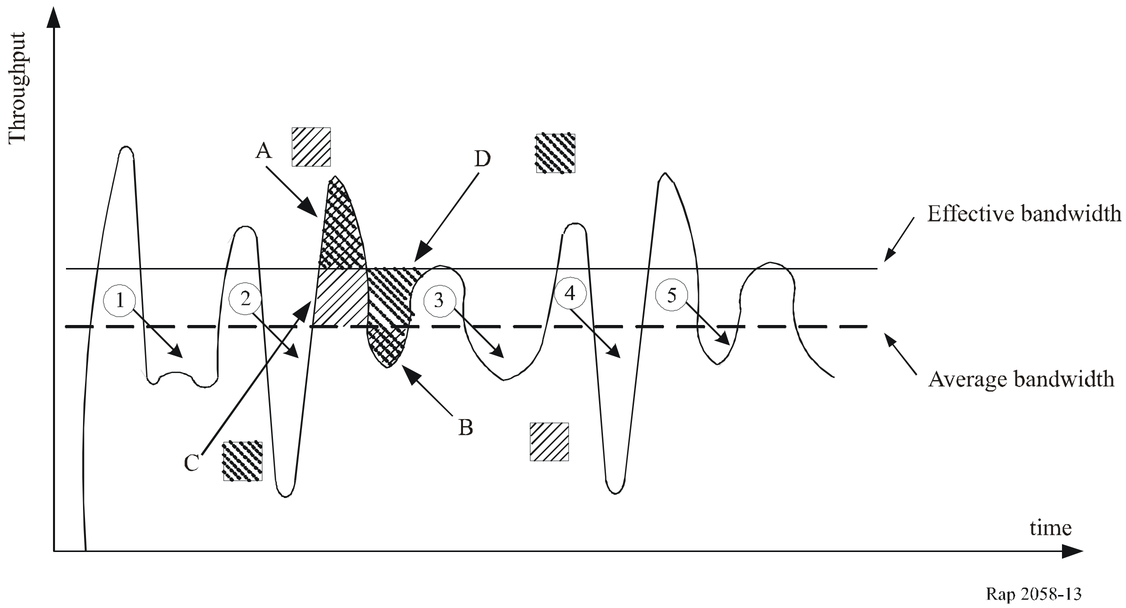
φ, from 0 to 1, is related to service delay requirement and decided by the operator. The lower the tolerant delay(a negotiated value) or the more frequent the traffic bursts, the larger φ should be decided, which means that more reserved resources are required, and higher QoS reliability can be obtained. In other words, high reliability is achieved by sacrificing the number of users admitted at the same time. Of course, if the required average throughput equals to the maximum one, φ is no longer used as in conversational service, and the equation (1) can be simplified to the equation (2)

$$Bandwidth_{effective} = Th_{average} = TH_{max} \tag{2}$$

How to choose a right value for φ is further discussed as below. As Fig. 13 indicated, packet data traffic is characteristic of burstiness and randomness. If the average throughput is set as effective bandwidth, a major part of data can be transmitted, during a small enough interval, even if the transient rate is higher than effective bandwidth. But peak 1 and peak 5 could not be sent in time, and would cause congestion or be rejected, which would affect system performance. Hence, in this case, the effective bandwidth which is larger than the average bandwidth should be chosen for those services with frequent burstiness or high priority.

FIGURE 13

Effective bandwidth and average bandwidth



Streaming service

The streaming class consists of real-time applications that send information to a viewer or listener, but without having any human response. Examples of this include video-on-demand, news streams, and multicasts.

Because of the absence of interaction, there is no longer a strict need for low delay, but the requirements for low jitter and media synchronization remain. Unlike voice service, streaming service is not CBR service and its max throughput value is usually larger than its average throughput value. The effective bandwidth of streaming service should be calculated as the following equation:

$$Bandwidth_{effective} = Th_{average} + \varphi * (TH_{max} - TH_{average}) \tag{3}$$

The parameters in equation (3) are defined the same as in equation (1). The rate of such kind of service does not change so greatly that φ can be determined primarily based on priority class.

For different φ, equation (3) may be simplified as

$$Bandwidth_{effective} = Th_{average} \qquad \text{for} \quad \varphi = 0 \qquad (4)$$

$$Th_{average} < Bandwidth_{effective} < TH_{max} \qquad \text{for} \quad 0 < \varphi < 1 \qquad (5)$$

$$Bandwidth_{effective} = TH_{max} \qquad \text{for} \quad \varphi = 1 \qquad (6)$$

Interactive service

This class covers a large variety of services that may differ greatly from each other in throughput and delay requirement, such as some games, network management system polling for statistics, and people actively web-browsing or searching databases. And the need for delay is reasonably prompt for the human activities, but not as low as for the conversational class.

Consequently, uniform bandwidth reservation for all kinds of interactive services would cause low usage of radio resource. So more factors, such as priority class, reliability class, burstiness as well, should be considered when calculating effective bandwidth. Then the effective bandwidth of interactive services can be estimated by the following equation.

$$Bandwidth_{effective} = \theta_1 * \theta_2 * Th_{average} + \theta_1 * \theta_2 * \varphi * (TH_{max} - TH_{average}) \qquad (7)$$

where θ_1, from 0 to 1, is the parameter based on priority class, and θ_2, from 0 to 1, is the parameter based on reliability class. And the above two values should be decided by the operator.

Background service

Background services covers all applications that either receive data passively or actively request it, but without any immediate need to handle the data. Examples of this include emails and file transfers.

Background services are not sensitive to delay, therefore several levels of average throughput or effective bandwidth R_i can be set according to resource condition of the FWA system, and then effective bandwidth of some service could be selected by service priority grade.

$$BW_{effective} = \begin{cases} R_1 \text{ (Priority 1)} \\ R_2 \text{ (Priority 2)} \\ R_3 \text{ (Priority 3)} \end{cases} \qquad (8)$$

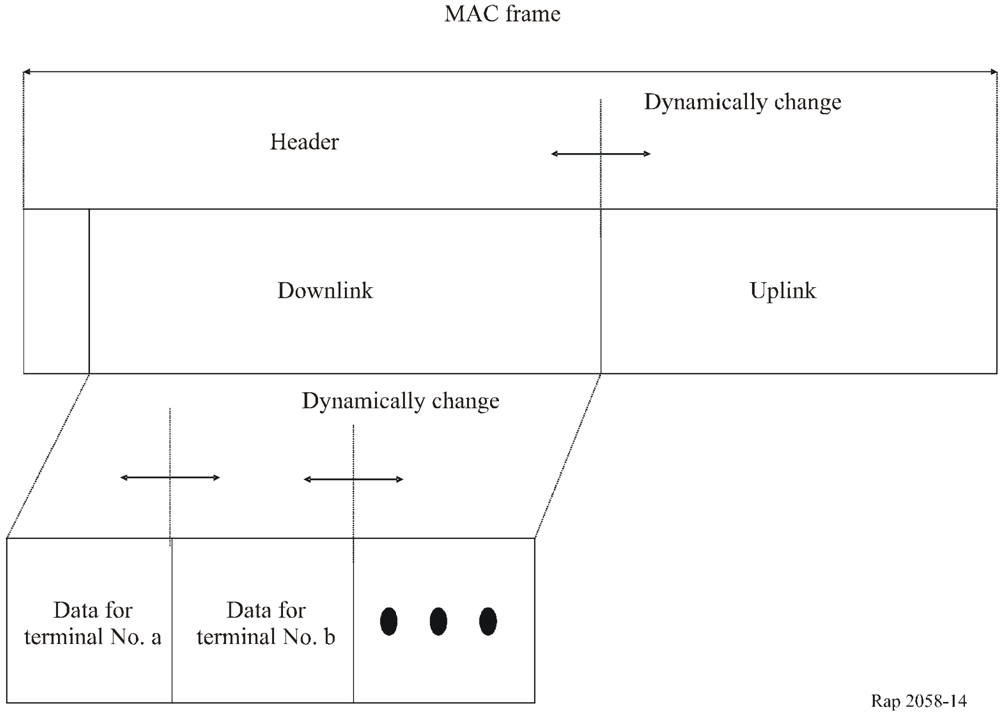
5.4 Scheduling function for a bandwidth assignment in the base station

5.4.1 Dynamic slot assignment (DSA)

A base station using the centralized control scheme assigns a specific bandwidth to each connection between itself and subscriber stations. Actually the base station assigns a TDMA time slot of MAC frame dynamically so that the system can handle asymmetry data between downlink and uplink or burst traffic data adaptively. This access method is called DSA. Figure 14 shows a MAC frame configuration model of TDMA-TDD/DSA system. The channel assignment ratio of downlink to uplink data, and data for individual terminal dynamically change according to traffic conditions.

FIGURE 14

MAC frame configuration model of TDMA-TDD/DSA system

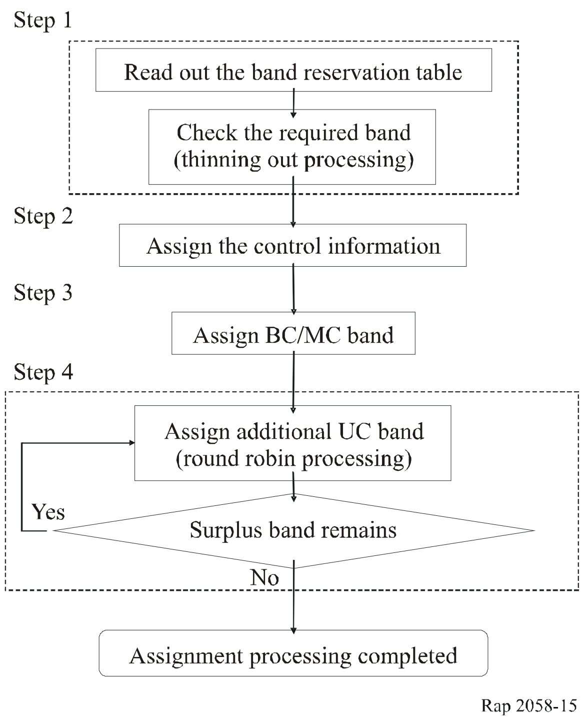


5.4.2 Bandwidth assignment control by scheduler

A scheduler in a base station assigns a bandwidth for individual connection between the subscriber stations according to the steps shown in Fig. 15.

FIGURE 15

Bandwidth assignment control by scheduler information data



Step 1: Assignment of the reserved bandwidth

The scheduler in the base station needs to enter the reserved bandwidth information on the bandwidth reservation table according to the required bandwidth and QoS class of each individual connection. On the bandwidth reservation table, the total reserved bandwidth in each frame is also managed. The scheduler assigns the number of data channels to each frame based on the reserved bandwidth information from an individual subscriber terminal.

Step 2: Assignment of the control

The control information required to maintain the wireless connection should be sent as speedily as required. To ensure the transmission of such control information, a scheduler in a base station sets the upper threshold of the bandwidth that can be reserved by each frame to send the control information stably.

Step 3: Assignment of broadcast (BC) and multicast (MC) bandwidth

If surplus bandwidth remains after Steps 1 and 2, a BC and MC bandwidth assignment is executed. When configuring MC connections, a scheduler calculates the maximum number of can-be-assigned data channels based on the bandwidth requested for each MC connection, and it then assigns the bandwidth within the upper limit. If the number of data channels to be sent exceeds the upper limit, the scheduler suspends the bandwidth assignment for a given interval. In this way, the scheduler can guarantee that BC or MC do not monopolize the bandwidth.

Step 4: Assignment of the additional bandwidth of each connection

If a surplus bandwidth remains in each frame after Steps 1 to 3, a scheduler assigns the additional bandwidth in a round robin manner. This does not depend on QoS class and the bandwidth is assigned to all connection equally. Here the scheduler sets the upper threshold of the number of data channels that can be given for the additional bandwidth. Data exceeding the threshold will be suspended until the next round robin assignment. This prevents any mobile terminal from monopolizing the bandwidth.

6 Comparison of QoS characteristics of FWA systems using a distributed control scheme and a centralized control scheme

A comparison of two systems is shown in Table 6.

TABLE 6

A comparison of QoS characteristics of FWA systems using a distributed control scheme and a centralized control scheme

	Distributed control scheme	**Centralized control scheme**
Type of QoS	Best effort	Guaranteed/Best effort
Merit	Affinity with Ethernet	1) Minimum bandwidth guarantee 2) Adaptive band allocation
Examples of systems	IEEE 802.11	ETSI-BRAN HIPERACCESS ETSI-BRAN HiperLANETSI-BRAN HIPERMAN MMAC-HSWA HiSWAN IEEE 802.16-2004

Annex 2

Example calculations of average access protocol delay and delay variation of CSMA/CA-based FWA

1 Introduction

In this Annex, theoretical method for estimating access protocol delay of RLAN using pure CSMA/CA technique (not using enhanced access control such as HCF (Hybrid coordination function) contention-based channel access for QoS support) is presented. The examples of calculation results in short packets flow situation and long packets flow situation are also described.

2 Approximations

Throughput per station (S_m), and access protocol delay ($\overline{T_m}$) are obtained by the following:

$$S_m = \frac{L_{ip} \times 8}{\overline{T_m} + SIFS + T_a} \tag{9}$$

$$\overline{T_m} = \sum_{n=0}^{\infty} (1 - p_m) p_m{}^n (A + nB) = A + B \frac{p_m}{1 - p_m} \tag{10}$$

where:

m: number of stations trying to send packets

L_{ip}: IP packets length (bytes)

T_d: data frame length

T_a: Ack frame length.

To derive the above approximations (9) and (10), the followings are being assumed.

– In case that the channel is busy, any random number within the same convention window (CW_{min}) is created, and the Backoff timer restarts.

– In case that transmitted packet collides with other packets from other stations, any random number within the same contention window CW_{min} is created and set to the backoff timer. Namely, BEB is not being considered.

The following § 3 and 4 describe examples of calculation results on access protocol delay of CSMA/CA (54M, 24M, and 6M mode) and CSMA/CA (11M, Long Preamble).

3 Examples of packet delay calculation in short packet flow situations

Assuming that multiple SSs attempt to send homogeneous short-length packets like VoIP packets simultaneously, the packet transmission delays are calculated. Used values for the parameters are listed in Table 7.

Figure 16 shows the average access protocol delay ($\overline{T_m}$) to be expected. $\overline{T_m}$ depends on transmission mode and the number of stations trying to send packets simultaneously, m. Here, the point to be paid attention is that the result in Fig. 16 does not consider BEB at the time when collisions occur. Because of that, the actual values of $\overline{T_m}$ will more degrade than the calculated values, especially when m becomes large and collisions increase.

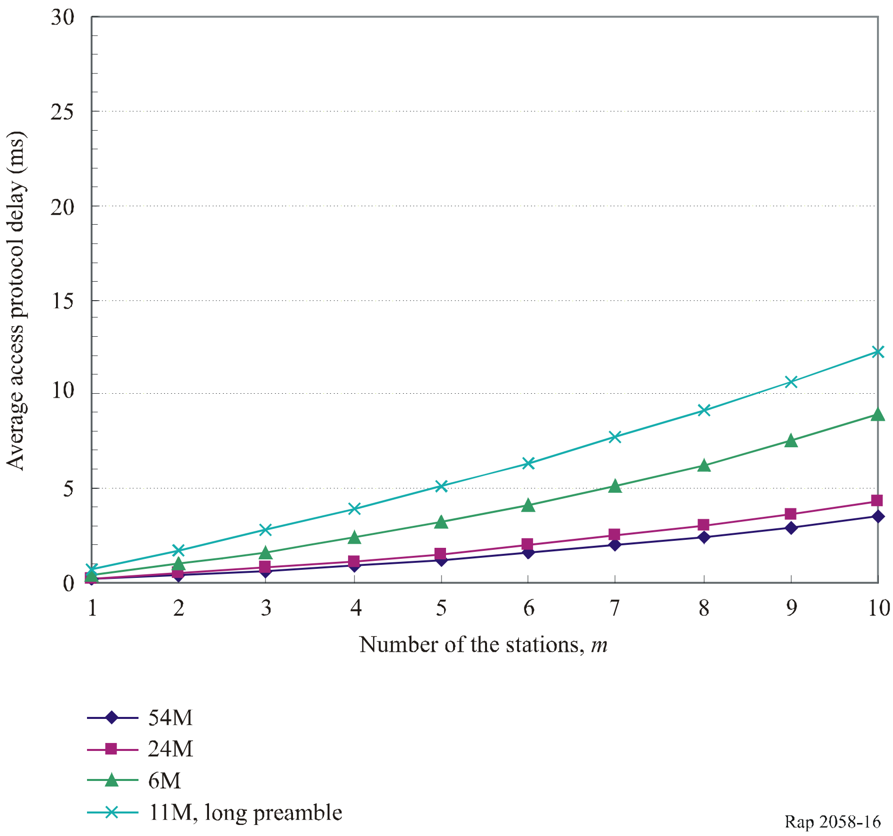
TABLE 7

Used values for parameters

	CSMA/CA (54M, 24M, 6M mode)	**CSMA/CA (11M mode, long preamble)**
SIFS (μs)	16	10
DIFS (μs)	34	50
SlotTime (μs)	9	20
CWmin	15	31
Packet length (bytes)	200	200
Td (μs)	56 (54M) 100 (24M) 340 (6M)	364
Ta (μs)	24 (54M) 28 (24M) 44 (6M)	202
Propagation error	None	None

FIGURE 16

Average access protocol delay of CSMA/CA (54M, 24M and 6M mode) and CSMA/CA (11M, long preamble) versus the number of stations in short packet flow situations



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4 Examples of packet delay calculation in long packet flow situations

Equations (9) and (10) are given under the condition that each station tries to send same size packets. Here, the key point to be paid attention is the access protocol delay that short packets like VoIP suffer on the condition that other multiple stations try to send long packets. In case that *m-1* stations try to send long size packets of same length and *one* station tries to send a short packet simultaneously, equation (10) should be modified for delay calculation. Modified equation (10) is as follows.

$$\overline{T_{m(S)}} = \sum_{n=0}^{\infty} (1-p_m) p_m{}^n \left(A_{(S)} + nB_{(L)}\right) = A_{(S)} + B_{(L)} \frac{p_m}{1-p_m} \qquad (11)$$

where:

$\overline{T_{m(S)}}$: Average access protocol delay for short packet transmission

$$A_{(S)} = DIFS + CW_{\min} \times SlotTime/2 + T_{d(S)}$$

$$B_{(L)} = T_{d(L)} + SIFS + T_a + DIFS + CW_{\min} \times SlotTime/2$$

m: the number of stations trying to send packets ($=m_{(L)} + m_{(S)}$)

$m_{(L)}$: the number of stations trying to send long packets

$m_{(S)}$: the number of stations trying to send short packets (=1).

Parameter values used for equation (11) are listed in Table 8. Other parameter values not shown in the Table are the same as ones in Table 7.

TABLE 8

Used values for parameters

	CSMA/CA (54M, 24M, 6M mode)	**CSMA/CA (11M mode, long preamble)**
Short packet length (bytes)	200	200
Long packet length (bytes)	1 500	1 500
Td (L) (µs)	248 (54M) 536 (24M) 2 072 (6M)	1 309
Td (S) (µs)	56 (54M) 100 (24M) 340 (6M)	364

Figure 17 shows the average access protocol delay ($\overline{T_{m(S)}}$) to be expected for a short packet transmission. $\overline{T_{m(S)}}$ depends on transmission mode and the number of stations trying to send packets simultaneously ($m=m_{(S)}+m_{(L)}$, here, $m_{(S)}=1$).

Comparing with Fig. 16, the access protocol delay in Fig. 17 becomes larger because the channel occupation time by other packet is much greater.

Figures 18 and 19 show the accumulated percentage of time on access protocol delay of CSMA/CA (54M mode) and CSMA/CA(11M mode, long preamble) when $m = 3$, 5 and 8 respectively. 1×10^{-3} quantile of access protocol delay $T_{m(S),1e-3}$ can be calculated by the following:

$$T_{m(S),1e-3} \approx A_{(S)} - B_{(L)}\left(\frac{3}{\log p_m}+1\right) \tag{12}$$

Calculated values are listed in Table 9.

TABLE 9

1×10^{-3} quantile of access protocol delay

Notes	CSMA/CA (11M mode, long preamble)	CSMA/CA (54M mode)	
$m_{(L)} = 2$, $m_{(S)} = 1$	33.5 ms	7.5 ms	$m = 3$
$m_{(L)} = 4$, $m_{(S)} = 1$	65.9 ms	15.8 ms	$m = 5$
$m_{(L)} = 7$, $m_{(S)} = 1$	122.1 ms	32.2 ms	$m = 8$

FIGURE 17

Average access protocol delay of CSMA/CA (54M, 24M and 6M mode) and CSMA/CA (11M, long preamble) versus the number of stations in long-packet flow situations

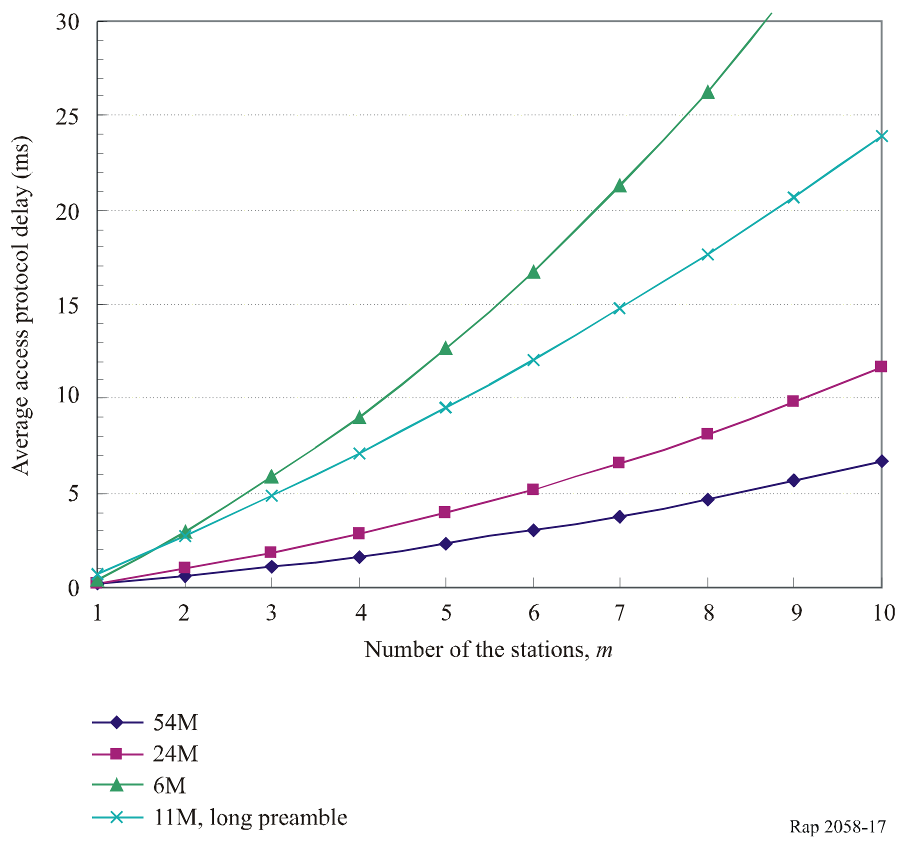

FIGURE 18

Accumulated percentage of time of access protocol delay (CSMA/CA, 54M mode)

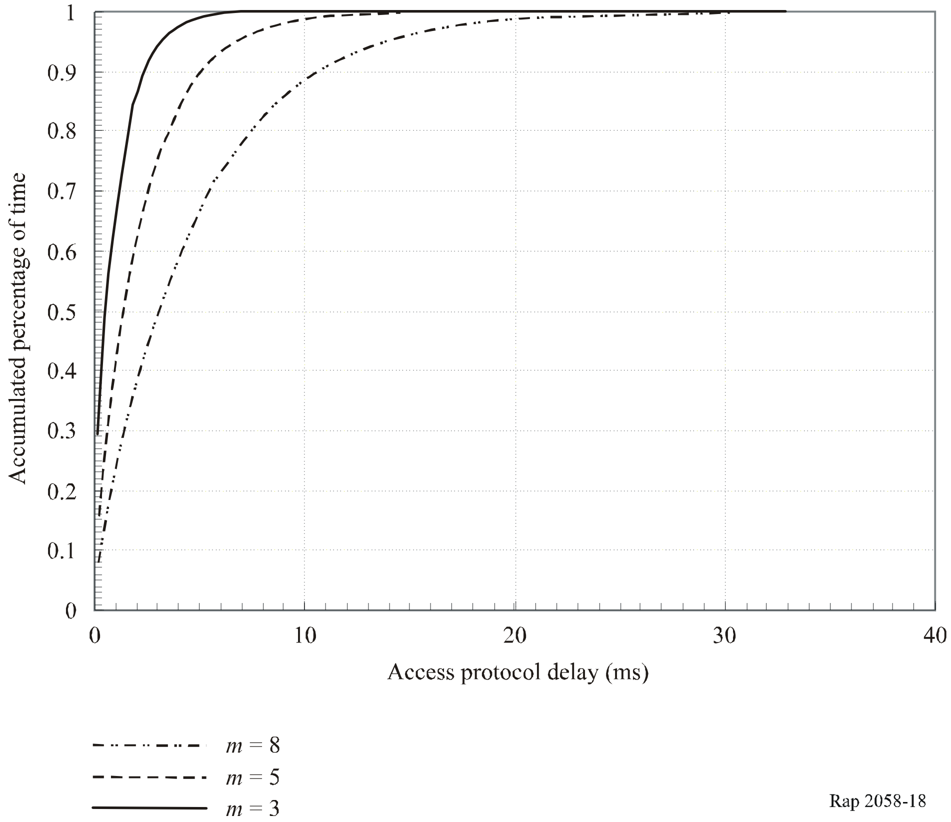

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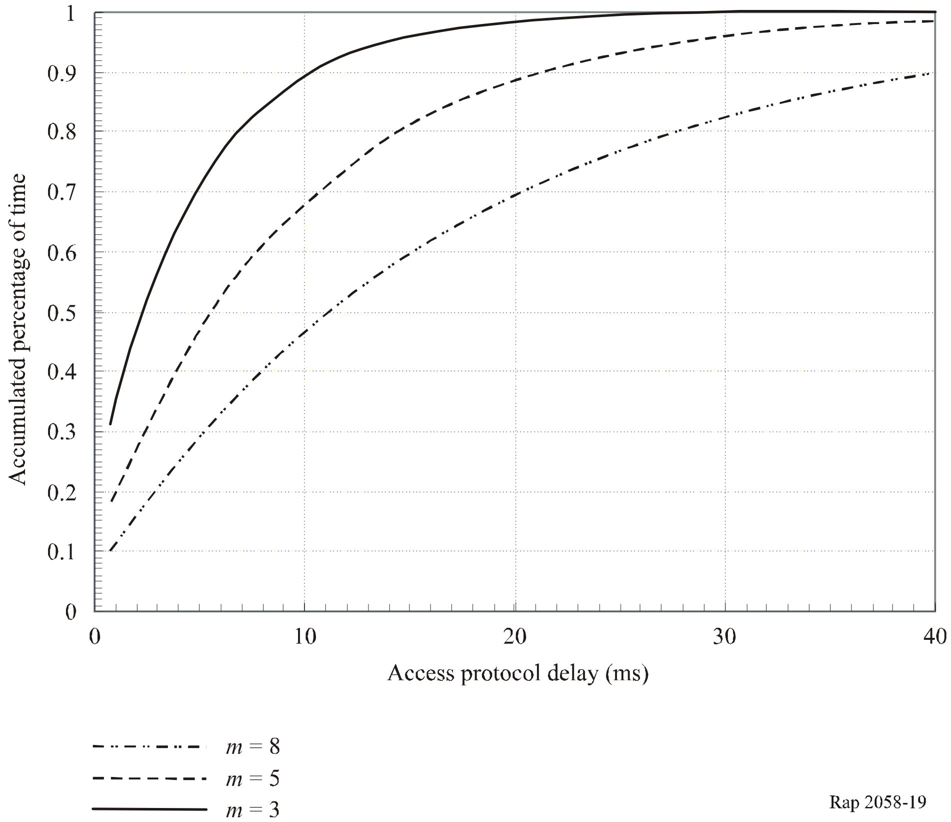
FIGURE 19

Accumulated percentage of time of access protocol delay (CSMA/CA, 11M, long preamble)



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Annex 3

Example calculations of additional waiting time in multiple VoIP flow situations for TDMA-based FWA

This Annex provides estimation of possible additional waiting time in multiple VoIP flow situations, referring to IEEE 802.16-2004 standard.

1 Uplink scheduling services

In § 1, scheduling services are designed to improve the efficiency of the poll/grant process. By specifying a scheduling service and its associated QoS parameters, the BS can anticipate the throughput and latency needs of the uplink traffic and provide polls and/or grants at the appropriate times.

The basic services are unsolicited grant service (UGS), real-time polling service (rtPS), non-real-time polling service (nrtPS) and BE service. Each service is tailored to a specific type of data flow. UGS is designed to support real-time service flows that generate fixed size data packets on a periodic basis, such as T1/E1 and Voice over IP without silence suppression. The rtPS is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as MPEG video. The nrtPS is designed to support non-real-time service flows that require variable size data grant burst types on a regular basis, such as high bandwidth FTP. The intent of the BE service is to provide efficient service to BE traffic.

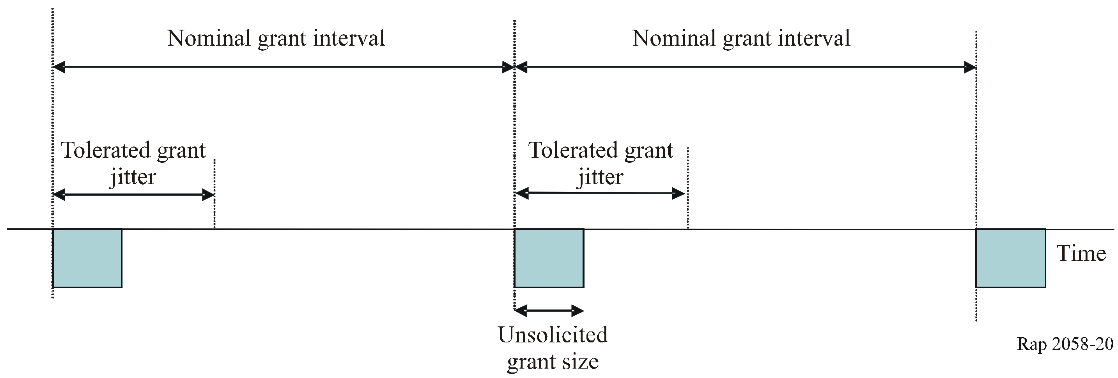
2 Unsolicited grant service (UGS)

As a scheduling service for VoIP packet transmission, we consider UGS. This service offers fixed size grants on a real-time periodic basis, which eliminate the overhead and latency of subscriber station (SS) requests and assure that grants are available to meet the flow's real-time needs. The BS should provide fixed size data grant burst types at periodic intervals to the service flow.

The UGS should be specified using the following parameters: the unsolicited grant size, the nominal grant interval, the tolerated grant jitter, and the request/transmission policy. Figure 20 illustrates the concepts of such parameters. Actual jitter can be held down within the tolerated grant jitter negotiated in call set-up procedure.

FIGURE 20

Key parameters for UGS service flow

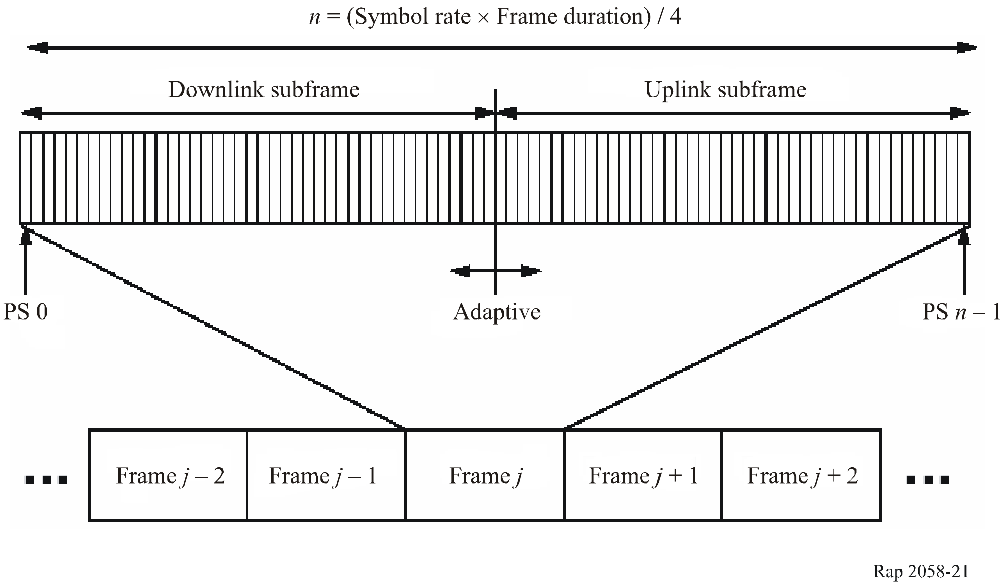


3 Frame structure

In the following study, we assume TDD as a duplex scheme. Figure 21 shows the frame structure in case of TDD. The frame has a fixed duration and contains one downlink and one uplink subframe. The frame size is usually 1 ms. The frame is divided into an integer number PSs, which help to partition the bandwidth easily. One PS is composed by 4 symbols in PHY layer. The TDD framing is adaptive in that the bandwidth allocated to the downlink versus the uplink can vary. The split between uplink and downlink is a system parameter and is controlled at higher layers within the system. Each physical channel size is in units of mini-slots. One mini-slots contains i PSs, where $i = 2^k$, and k is an integer ranging from 0 through 7.

FIGURE 21

TDD frame structure



4 Assumptions for calculation

In the multiple VoIP flow situation, packets to be sent of newly-generated VoIP flow may wait due to channel occupations by preceding other VoIP flows. We calculate an example of the additional waiting time in the situation. Assumed parameter values are listed in Table 10.

The calculation was conducted based on the following assumptions.

– All VoIP coders are compliant with ITU-T Recommendation G.711 (64 kbit/s coding). Considering TCP/IP header, Ether header, MAC header, and so on, MAC PDU length is totally 234 bytes. Assuming outer Reed Solomon coding and inner BCC, the burst length becomes 381 bytes. Assuming QPSK modulation, symbol size after mapping and preamble appending is 1 540 in total.

– QPSK modulation and 25 MHz channel size are assumed. Frame duration is 1 ms recommended. Thus, 20 000 symbols contain in the frame duration.

– Basically uplink and downlink VoIP traffic loads are the same, but we assume that a BS handles rtPS, nrtPS, and BE as well as VoIP traffics simultaneously, and some services in those may have a nature of asymmetric traffic load that the downlink load is larger than the uplink load, such as MPEG streaming. Thus, we assume that the ratio of uplink/downlink subframe length is about 1:3. Thus, the number of symbols assigned for uplink subframe is assumed to be approximately 5 000.

– Considering assignment of other kinds of real-time services and non-real-time services to the uplink, the number of VoIP bursts assigned to the uplink is about 2 at the maximum. Fragmentations of VoIP packets are not considered.

TABLE 10

Assumed parameter values

		Note
MAC PDU size	234 bytes	
– VoIP payload	160 bytes	G.711-compliant
– TCP/IP header	40 bytes	
– Ether header	24 bytes	
– FCS	2 bytes	
– PHSI	2 bytes	
– MAC header	6 bytes	
Modulation	QPSK	
FEC code type	Reed Solomon and BCC	
Outer code type	Type 2	
Total outer codeword size (K+R)	254 bytes	
Inner BCC code type	(24,16)	
Preamble length	16 symbols	
Roll-off factor	0.25	
Channel size	25 MHz	
Symbol rate	20 Mbaud	
Bit rate	40 Mbit/s	
Frame duration	1 ms	
Number of symbols within 1 frame	20 000	
Number of minislots within 1 frame	5 000	Mini-slot size is equal to PS size (4 symbols).

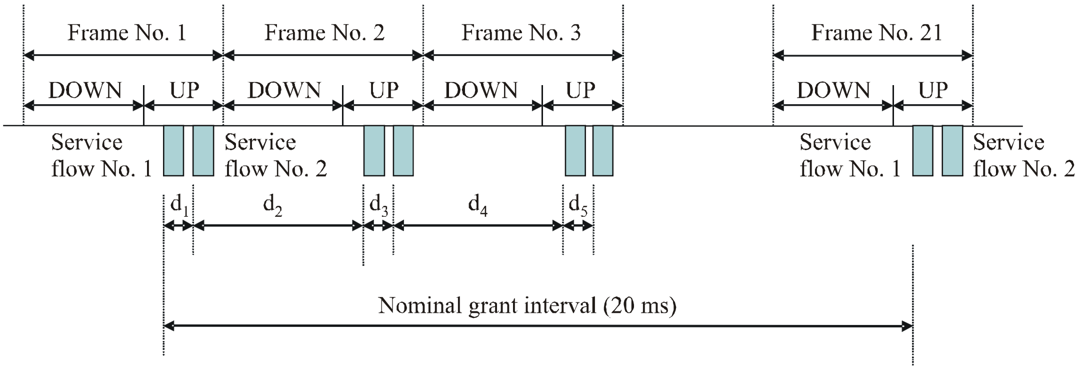
– VoIP packets are generated at 20 ms interval. To suppress jitters between VoIP packets, bursts of one VoIP flow are transmitted using same time slot positions in uplink subframes at intervals of 20 ms.

– On the above assumptions, as illustrated in Fig. 22, 40 grants at the maximum can be assigned for VoIP flows at 20 ms intervals. Using one grant of those, an uplink burst of each VoIP flow is transmitted.

– In case that preceding VoIP flows do not exists, an uplink burst of a newly-generated VoIP flow will be transmitted at time t_0 using the nearest grant on a time-axis. On the other hands, in case that one or more VoIP flow(s) already exist(s) and the grants are reserved for existing VoIP flow(s), an uplink burst of a newly-generated VoIP flow may be unable to use the nearest grant, and a BS scheduler will search a non-reserved grant and serve it to the SS. The SS gets information from UL_MAP in downlink subframe, and sends the burst at time t_1. Namely additional wait (t_1 - t_0) may occur occasionally. In the following, such kind of waiting time is calculated.

FIGURE 22

VoIP grant



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– Now, we assumed that $(m - 1)$ VoIP flows are being served. Average additional waiting time occurred in new *mth* VoIP flow $\overline{D_m}$, can be approximated as the following.

$$\overline{D_m} \approx (N - m + 1) \cdot \frac{(m-1)!}{N!} \cdot \sum_{j=1}^{m-1} \left\{ \sum_{i=1}^{j} d_i \cdot \frac{(N-j-1)!}{(m-j-1)!} \right\} \tag{13}$$

$$1 \le j \le m-1$$

where:

N: Possible maximum number of VoIP grants within 20 ms time period (= 40)

d_i: Time difference between $(i - 1)$-th and i-th grant.

In addition, for simple calculation,

$d_{odd} = 0.08$ ms, $d_{even} = 0.92$ ms

are assumed.

5 Results of example calculations

Calculation results of $\overline{D_m}$ approximated by equation (13) is shown in Fig. 23. Simulation results are also shown in the Figure. Depending on the number of existing VoIP flows, the waiting time varies. Figure 24 depicts the accumulated percentage of the waiting time calculated by equation (13). Occasionally, 20 ms waiting may occur.

Under some assumptions, possible additional waiting time to be expected for VoIP packets in multiple VoIP flow situations was estimated. For TDMA, scheduling algorithm is a key factor to improve delay performance, but it is out of scope of IEEE802.16-2004 standard. In addition to such algorithm, several factors like frame structure, traffic loads and capacity may become causes of IPTD degradation.

FIGURE 23

Example of average additional waiting time that *m*-th VoIP flow suffers

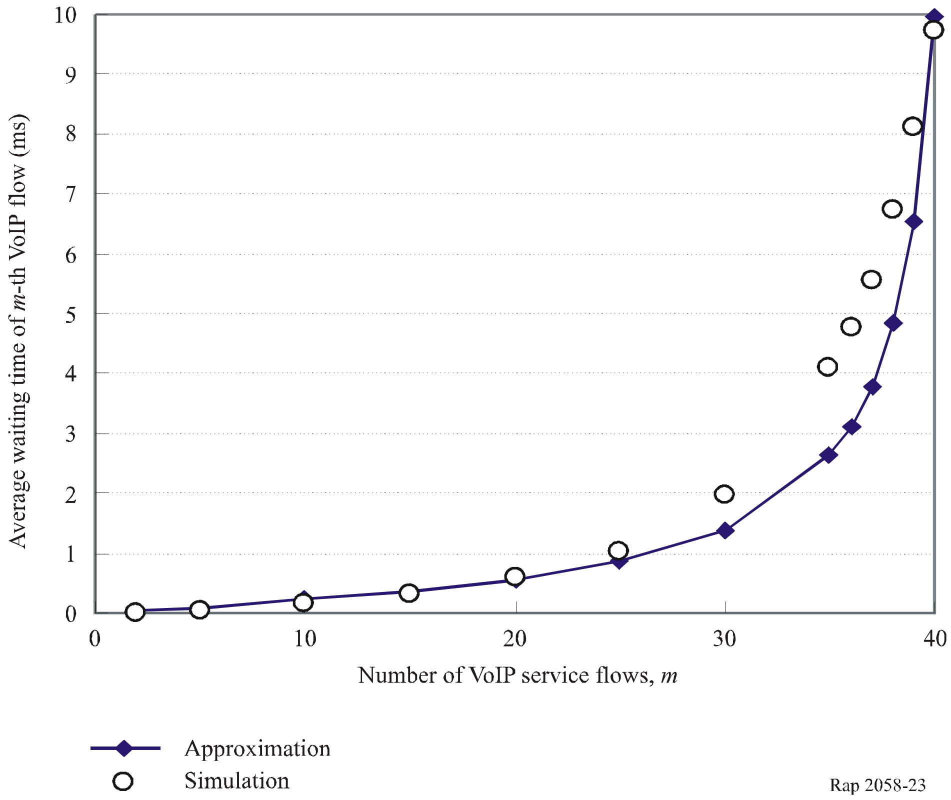
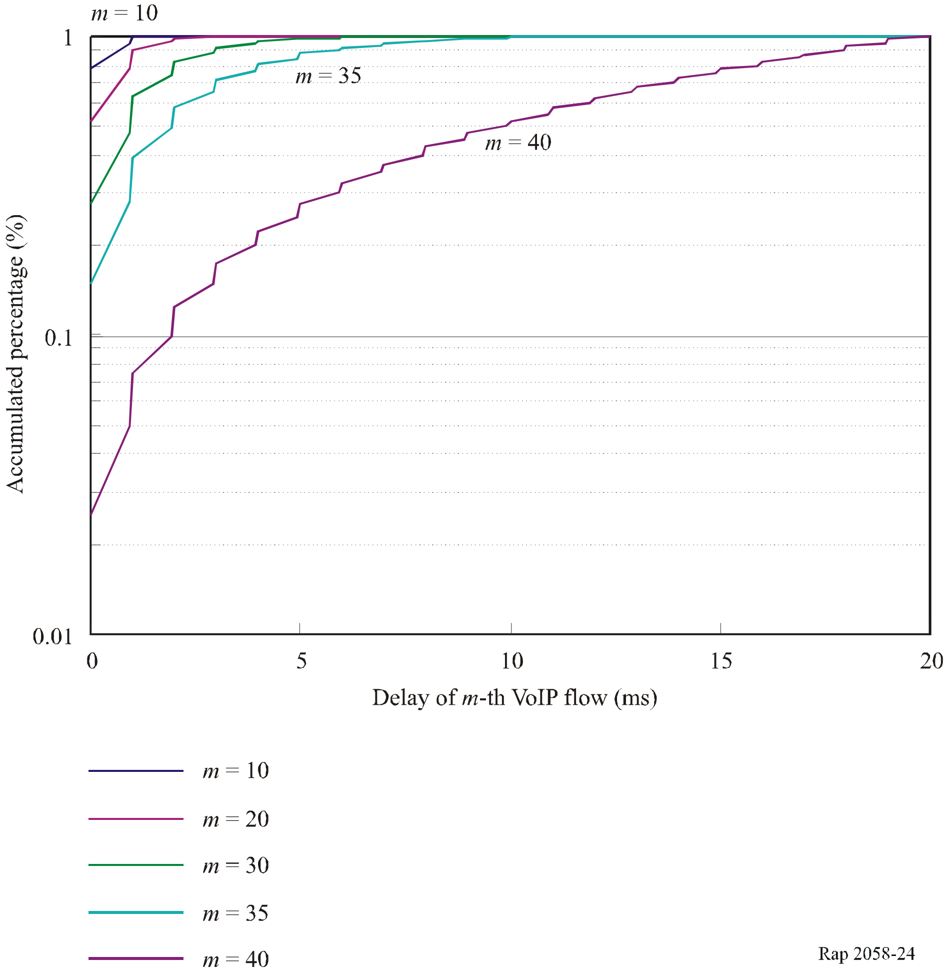


FIGURE 24

Example of accumulated percentage of additional waiting time that *m*-th VoIP flow suffers



Annex 4

Example calculations of QoS Class 0 network delay

This Annex presents example calculations of IPTD for any path portion supporting a QoS Class 0 flow based on the methodology described in Appendix III to ITU-T Recommendation Y.1541.

1 Delay calculation in access networks including FWA systems

In this paragraph, concept of delay calculations in access networks including FWA systems is described. Figure 25 shows access network configuration including FWA system. IPTD and IPDV in the access network, $D(AN)$ and $DV(AN)$, are calculated in the following equations.

$$D(AN) = D(FWA) + D(BH)$$

$$= D(s) + D(air) + D(b) + D(BH)$$

$$DV(AN) \leq DV(FWA) + DV(BH)$$

$$= DV(s) + DV(air) + DV(b) + DV(BH)$$

As indicated in Fig. 25, the link between the base station and the access gateway is defined as the backhaul. The backhaul is the part of access network and made up of a fibre optic, coaxial, copper cable, or another radio system, for example a P-P or P-MP fixed wireless system. Usually the backhaul should be designed to have enough capacity to convey the traffic from/to the FWA system. Thus, IPTD and IPDV occurring in the backhaul, $D(BH)$ and $DV(BH)$ are usually small. As the result, the above equations are modified to;

$$D(AN) \approx D(FWA)$$

$$= D(s) + D(air) + D(b)$$

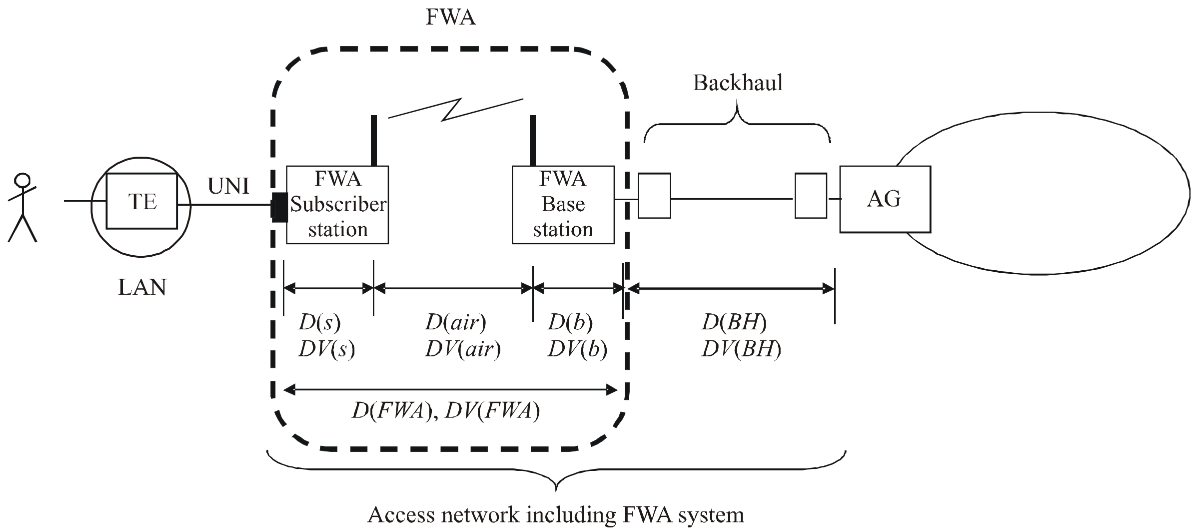
$$DV(AN) \leq DV(FWA)$$

$$= DV(s) + DV(air) + DV(b)$$

Furthermore, if the dominant cause of $D(FWA)$ and $DV(FWA)$ is the access protocol and other causes can be neglected, $D(AN)$ and $DV(AN)$ may be approximated roughly as $D(AN) \approx D(air)$ and $DV(AN) \approx DV(air)$.

FIGURE 25

Example of access network including an FWA system



TE: Terminal equipment
AG: Access gateway
FWA: Fixed wWireless access
UNI: User-network interface

$D(FWA)$: Delay time in the FWA portion
-$D(s)$: Delay time in the FWA Subscriber Station
-$D(air)$: Delay time depending on the access protocol
-$D(b)$: Delay time in the FWA Base Station
$D(BH)$: Delay time in the backhaul

$DV(FWA)$: Delay variation in the FWA portion
-$DV(s)$: Delay variation of the Subscriber Station
-$DV(air)$: Delay variation depending on the access protocol
-$DV(b)$: Delay variation of the FWA Base station
$DV(BH)$: Delay variation in the backhaul

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2 Example calculations of QoS class 0 network delay in UNI-UNI

According to ITU-T Recommendation Y.1541, the theoretical value of IPTD in IP network portion is:

$$IPTD\ (\text{ms}) \leq (R_{km} * 0.005) + (N_A * D_A) + (N_D * D_D) + (N_C * D_C) + (N_I * D_I)$$

$$R_{km} = 1.25 * D_{km}$$

In this equation:

– D_{km} is the air-route distance between two routers that bound the portion.

– R_{km} represents the route length assumption.

– N_A, N_D, N_C, and N_I represent the number of IP access gateway, distribution, core and internetworking gateway routers respectively; consistent with the network section example in Fig. III.1 in Appendix III to ITU-T Recommendation Y.1541.

– D_A, D_D, D_C, and D_I, represent the delay of IP access gateway, distribution, core, and internetworking gateway routers respectively; consistent with the values defined in Table III.1 in Appendix III to ITU-T Recommendation Y.1541.

Maximum IPDV may be calculated similarly.

As an example for UNI-UNI delay calculation, hypothetical reference path (HRP) illustrated in Fig. 26 is adopted. The following assumptions are considered.

– In two access portions, one portion consists of FWA and the backhaul link.

– The remaining portions, such as two IP networks and non-IP network which consists of the other access portion are consistent with the HRP illustrated in Fig. III.3 in Appendix III to ITU-T Recommendation Y.1541.

– The non-IP network has T1 capacity.

– Largest packet size is 1 500 bytes, and VoIP packet size is 200 bytes.

– Total distance and route length are 4 070 km and 5 087.5 km respectively, complying with the assumption in Appendix III to ITU-T Recommendation Y.1541.

Table 11 gives the hypothetical reference path (HRP) configuration in terms of number and type of routers, distance, and contribution of all HRP components to IPTD and IPDV in UNI-to-UNI. Except for values related to the access portion including FWA system, each component value included in IP Net 1, IP Net 2, and non-IP is conformed to Table III.2 in Appendix III to ITU-T Recommendation Y.1541.

IPTD and IPDV of access network including FWA system, $D(AN)$ and $DV(AN)$ are depending on the access protocol as described in section 1 in this Annex, and as described in § 6.3, they vary drastically according to some conditions like total traffics. Thus, it is difficult to fix values.

There are some examples of access protocol delay in case that a CSMA/CA-based wireless access protocol is adopted to the FWA system in Annex 2 in this Report. From the Annex, it is assumed that $D(AN)$ varies within the range of several milliseconds or several ten ms. In this analysis, 10 ms is adopted as an example. This value (10 ms) is also set as the performance objective of ETSI HIPERACCESS (see ETSI TR 101 177 V1.1.1[1]). Annex 2 also shows examples of 1×10^{-3} quantile of access protocol delay. Though 1×10^{-3} delay quantile also varies according to several conditions, 16 ms in case of CSMA/CA (54M mode, $m = 5$) is being adopted as an example.

FIGURE 26

Hypothetical reference path for QoS Class 0 (including FWA system)

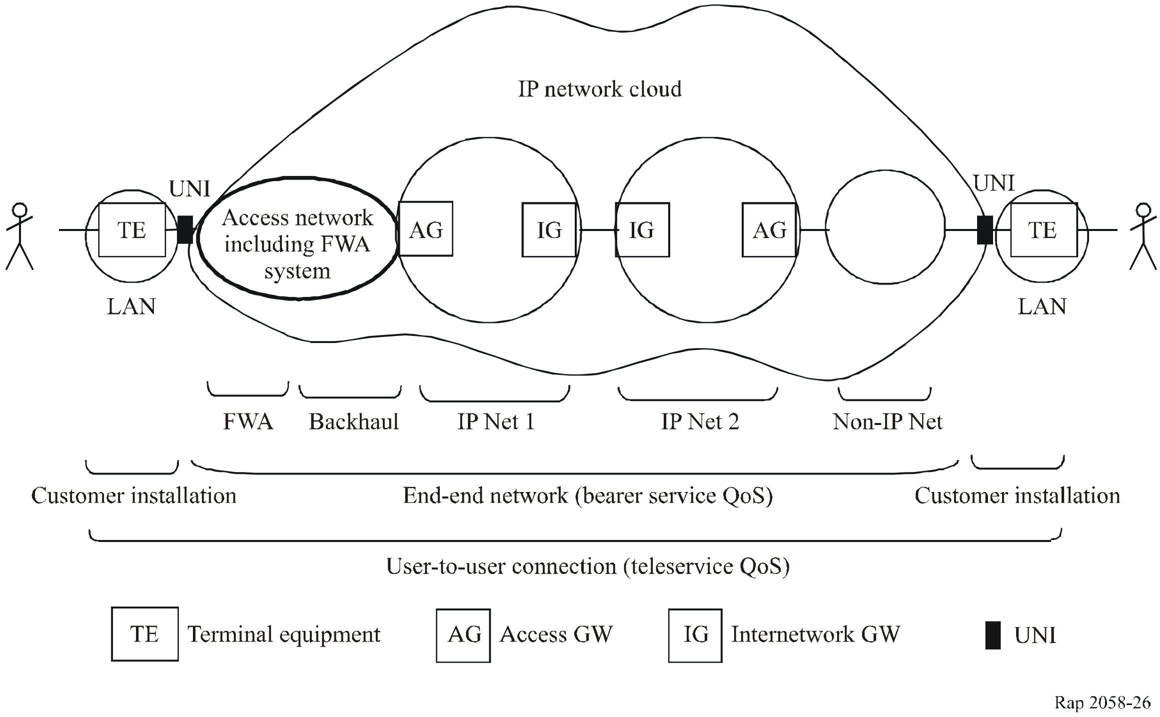


TABLE 11

Delay analysis of example QoS Class 0 path (in case of CSMA/CA as an FWA access protocol)

Element	Unit	IPTD/Unit (ms)	Average IPTD (ms)	IPDV/Unit (ms)	Max IPDV (ms)
Distance	4 070 km				
Route	5 087.5 km		25		
Access network including FWA system	1	1	**10**	1	**16**
IP Net 1			20		28
Access, N_A Distribution, N_D Core, N_C Internetwork GW, N_I	1 1 2 1	10 3 2 3	10 3 4 3	16 3 3 3	16 3 6 3
IP Net 2			24		34
Access, N_A Distribution, N_D Core, N_C Internetwork GW, N_I	1 1 4 1	10 3 2 3	10 3 8 3	16 3 3 3	16 3 12 3
Non-IP Net			15		0
UNI-UNI Total (ms)			**94**		**78**
ITU-T Recommendation Y.1541 specifications			100		50

The calculated value of IPTD in UNI-UNI is 94 ms, and is within specification of 100 ms for Class 0. On the other hand, the calculated value of IPDV in UNI-UNI is 78 ms, and this value is greater than the specification in ITU-T Recommendation Y.1541 (50 ms). In this respect, further studies will be needed. However, it must be noted that;

– 78 ms is calculated from 16 ms (DV(AN)) and 62 ms (IPDV due to other portions);
– 62 ms due to other portions is coincident with the value used for an example of UNI-UNI IPDV calculation in Table III.2 in Appendix III to ITU-T Recommendation Y.1541; and
– ITU-T Recommendation Y.1541 describes that 62 ms itself is very pessimistic, assuming worst case addition of each router.

Table 11 also shows a UNI-UNI delay example in case of CSMA/CA as an access protocol for FWA system. In FWA system adopting TDMA, IPTD and IPDV may be improved.

Annex 5

Technical characteristics of broadband FWA systems to support VoIP technique

1 Introduction

In recent years, VoIP service has been one of the most attractive applications for broadband internet service providers even including the conventional telecommunication operators. ITU-T has been standardizing several aspects of VoIP technique, such as protocols, transmission and voice qualities. There are several ITU-T Recommendations which define the end-to-end performance objectives for IP-based network. When VoIP service is provided by FWA system, performance objectives for the wireless section should be also defined to match the end-to-end performance objectives. This Annex provides considerations to satisfy the performance parameters for VoIP conveyed in FWA systems.

2 Required parameters for VoIP in relation to the end-to-end performance objectives

Performance objectives for IP packet transmission are provided in ITU-T Recommendations Y.1540 and Y.1541. ITU-T Recommendation Y.1540 defines availability performance parameters, and Y.1541 specifies six QoS class and IP packet transfer performance parameters. Both Recommendations are applied to end-to-end, international IP data communication service. They support a wide range of IP application such as VoIP, multimedia conferencing or interactive data transfer. Therefore, some parts of performance objectives are enough for voice transmission.

There are two schemes for measuring voice quality for telephony. One is subjective scheme and the other is objective scheme. The mean opinion score (MOS) evaluation is generally used in the subjective tests, which are mathematically averaged to obtain a quantitative indicator of the network performance. ITU-T Recommendation P.800 provides detailed MOS measurement procedure. The PESQ evaluation is one of representative objective tests, which provides the measurement of clearness by comparing an input test signal with the signal output across the network. ITU-T Recommendation P.862 provides PESQ measurement procedure.

PESQ alone cannot ensure comprehensive voice quality. E-model, which is provided by ITU-T Recommendation G.107, is used to fully evaluate voice quality. It is used to calculate the transmission rating factor *R*. *R* is calculated as follows:

$$R = R_0 - I_s - I_d - I_{e,\,eff} + A \qquad (14)$$

where:

R_0: basic signal-to-noise ratio,

I_s: combination of all impairments which occur simultaneously with the voice signal,

I_d: impairment due to delay,

$I_{e,\,eff}$: impairment due to low bit rate coding and packet loss.

The advantage factor *A* allows for compensation of impairment factors such as mobile communication.

The TTC standard JJ-201.01 provides these parameters, which are used with *R*, and their evaluation method.

3 Consideration to satisfy the performance parameters for VoIP conveyed in FWA systems

Performance parameters which have impact on voice quality estimated by *R* are delay and echo. Packet loss, which is one of distinctive parameter in packet-based network, also affects the voice quality. These performance parameters vary as a function of the wireless system.

3.1 Delay and echo

Delay and echo are parameters to be considered seriously in wireless systems. There are three delay parameters described in E-Model as shown in Table 12.

TABLE 12

Delay parameters in E-Model

Parameter	Abbreviation	Unit	Range
Absolute delay in echo-free connections	*Ta*	ms	<500 ms
Round trip delay of the echo path	*Tr*	ms	<1 000 ms
Mean one-way delay of the echo path	*Ta*	ms	<500 ms

Ta is too-long absolute delay, which occurs even with perfect echo cancelling. *Tr* and *T* cause listener echo and talker echo respectively. When FWA system provides VoIP service, packet transfer delay in section of wireless access should be less than 500 ms minus delay of standard terminal including coders, core network and wire access network in the opposite side. When wireless systems are used for both access network or multi hop wireless access network is used, the delay in each wireless access section should be design as the total amount of delay in wireless access sections satisfies the objective. This delay objective is a provisional value because it is maximum delay used in E-model. Further study to decide precise objective is required.

3.2 Packet loss

Evaluation of packet loss is necessary for voice quality evaluation. Packet loss increases when network conditions are such that packet traffic exceeds the designed network performance capabilities. Such condition occurs in both wired and wireless systems. Packet loss probability (P_{pl}) is larger in wireless systems than wired systems because network performance in wireless systems is usually inferior to wired systems. In wireless systems, the data transmission rate is lower than wired (optical) systems and the signal may sometimes degrade due to interference, shadowing or rain attenuation.

P_{pl} is composed of the packet loss probability in network (P_{plN}) and the packet loss probability in a jitter absorption buffer (P_{plB}). P_{plN} is the ratio of the number of received packets over the number of transmitted packets. P_{plB} is the wasted packets caused by the overflow of the jitter absorption buffer. When an FWA system provides VoIP service, the packet loss ratio should be less than 20% minus P_{plB} and P_{plN} of a wired network as shown in Table 13. Similar to delay objective, when wireless systems are used for both access or multi hop wireless access network is used, the packet loss probability in each wireless access network should be designed as the whole parts of wireless access network to satisfy the objective. This is also a provisional value because it is maximum value of random packet-loss probability used in E-model. Further study to decide precise objective is required.

TABLE 13

Packet loss parameter

Parameter	Abbreviation	Unit	Range
Random packet-loss probability	P_{pl}	%	<20

The calculation model and method of P_{plB} are illustrated in TTC Standard JJ-201.01 and "Network performance metrics in estimating the speech quality of VoIP" (MASUDA, M. and ORI, K. [November, 2001] IEICE APSITT2001, p. 333-337) respectively.

4 Example of speech quality for hypothetical paths including FWA

In this paragraph, the processing delay of ITU-T Recommendation G.711-compliant voice coder is estimated. Then, based on the mouth-to-ear delay including UNI-UNI delay and coder delay analysis, the calculation results of R-value defined in ITU-T Recommendation G.107 for a parameter for speech quality evaluation is described.

4.1 Coder delay

Recently, several kinds of coders, such as ITU-T Recommendation G.711, G.729, and G.723.1 are used for VoIP. In those coders, most popular voice coding technique is ITU-T Recommendation G.711. Coded data is not compressed, and its coding rate is 64 kbit/s. Assumed values for delay components are listed in Table 14.

TABLE 14

Example of ITU-T Recommendation G.711-compliant coder delay

		Delay (ms)	Notes
Packet formation		40	2 times of the frame size
De-jitter buffer, average		40	In case of 80 ms buffer
Packet loss concealment	Appendix I/G.711	10	One PLC "frame"
	None	0	
Total		80 or 90	80 ms (without PLC), 90 ms (with PLC)

VoIP packet length is assumed to be 200 bytes containing 40-byte TCP/IP header and 160-byte payload. Then the time frame of the coder is calculated to be 20 ms. Thus, in accordance with ITU-T Recommendation G.113, the processing delay for packet formation is assumed to be the double of 20 ms, namely 40 ms. De-jitter buffer is used to absorb the jitter between incoming packets. And ideally, its buffer size is decided according to maximum IPDV. For example, if IPDV is 78 ms calculated in Table 11 in the Annex 4, the buffer size is assumed to be 80 ms. ITU-T Recommendation Y.1541 describes that de-jitter buffer's contribution is based on the average time that packets spend in the buffer, and it can be assumed to be half of the buffer size. Therefore in this case, de-jitter buffer's contribution is assumed to be 40 ms. In addition, in case of using the packet loss concealment (PLC) technique, 10 ms is added. As the result in this case, the coder delay is estimated to be 80 ms or 90 ms totally.

Total coder delay can be decreased by reducing the packet formation delay and/or the de-jitter buffer size. The packet formation delay depends on the frame size, and smaller frame size can improve the coder delay. However, ratio of VoIP header length to the total VoIP packet length becomes larger, and as the result, transmission efficiency of the link will be degraded. Smaller size de-jitter buffer can also decrease the coder delay. However, making buffer size smaller leads to increase of out-of-order packet arrival, and such packets are discarded.

4.2 Mouth-to-ear delay

In ITU-T Recommendation G.114, mouth-to-ear delay is recommended to be less than 150 ms. This paragraph shows an example of calculation result on mouth-to-ear delay. Mouth-to-ear delay depends on $D(AN)$ and $DV(AN)$, both of which vary with some conditions, such as FWA access protocol and the number of subscriber stations.

The following are assumed:

– Delay contributions to IPTD and IPDV (UNI-UNI) are based on Table 7 in the Annex 2, except for contributions by access portion including FWA system, namely D(AN) and DV(AN).

– The dominant factor causing D(AN) and DV(AN) is an access protocol delay of FWA system, and other factors are neglected.

– The FWA system adopts pure CSMA/CA-based wireless access protocol.

– On the condition that m–1 stations through 1st to $(m-1)$-th station attempt to send long packets (1 500 bytes) like FTP, the other *one* station (m-th station) tries to send a short packet (200 bytes) for VoIP.

– Delay contributions by coder delay processing is based on Table 9, except for a de-jitter buffer's contribution. The de-jitter buffer's delay contribution is assumed to be depending on total UNI-UNI IPDV. (The buffer size is assumed to be equal to UNI-UNI IPDV, and the average de-buffer's contribution is assumed to be half of the buffer size.)

Based on the above assumptions, mouth-to-ear delay $D(total)$ occurring at a short packet transmission can be roughly calculated by the following:

$CoderDelay = PacketFormation + DejitterBuffer + PLC$

$$PacketFormation = 40 \text{ ms}$$

$$DeJitterBuffer \approx \frac{DV(AN) + DV(IP\ Net\ 1) + DV(IP\ Net\ 2)}{2}$$

$$DV(AN) \approx DV(FWA) = A_{(S)} - B_{(L)}\left(\frac{3}{\log\ p_m} + 1\right) \qquad (15)$$

$$DV(IP\ Net\ 1) = 28 \text{ ms}$$

$$DV(IP\ Net\ 2) = 34 \text{ ms}$$

$$PLC = \begin{cases} 10 \text{ ms} & (\text{with } PLC) \\ 0 & (\text{without } PLC) \end{cases}$$

$$\begin{aligned} D(total) &= IPTD(UNI \cdot UNI) + CodertDelay \\ &= D(AN) + D(Route) + D(IP\ Net\ 1) + D(IP\ Net\ 2) + D(NonIP\ Net) + CoderDelay \end{aligned}$$

where:

$$D(AN) \approx D(FWA) \approx A_{(S)} + B_{(L)}\frac{p_m}{1-p_m} \qquad (16)$$

$D(Route) = 25$ ms

$D(IP\ Net\ 1) = 20$ ms

$D(IP\ Net\ 2) = 24$ ms

$D(NonIP\ Net) = 15$ ms

$A_{(S)}$, $B_{(L)}$, and equations (13) and (14) are given in Annex 5.

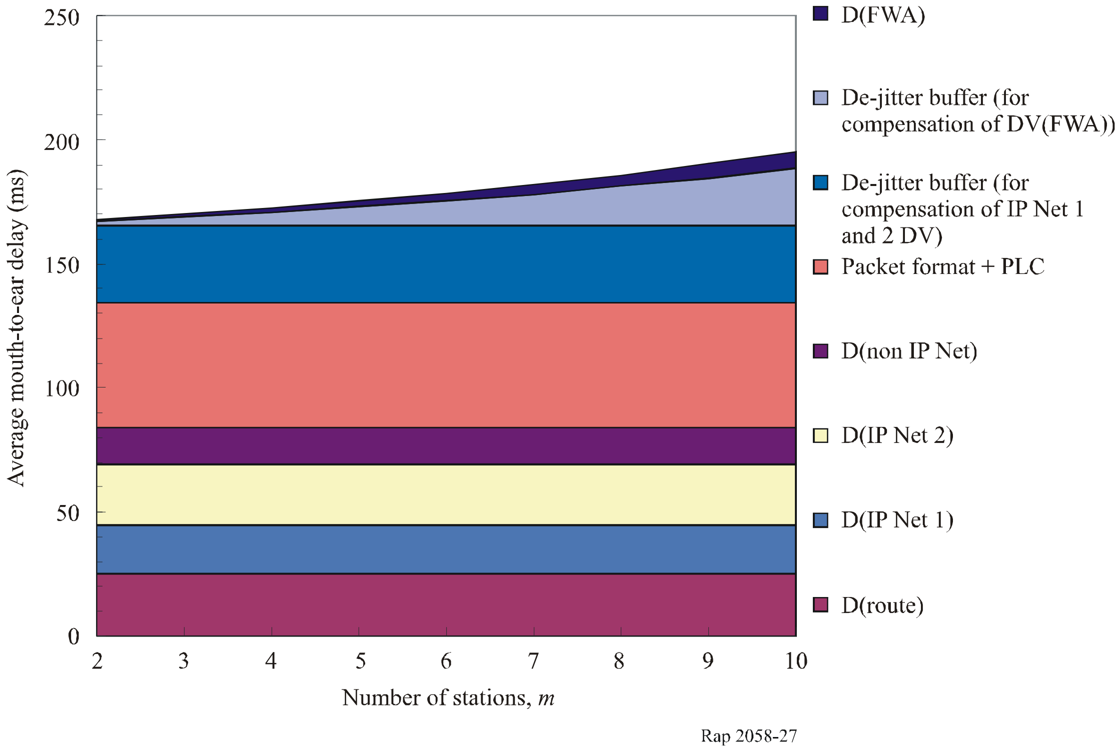
Therefore, $D(total)$ is calculated by the following:

$$D(total) \approx \frac{3}{2}A_{(S)} + \frac{B_{(L)}}{2}\left(\frac{2p_m}{1-p_m} - \frac{3}{\log p_m} - 1\right) + 165 \ (\text{or } 155)$$

Figure 27 depicts the example of calculated mouth-to-ear delay that a short packet from mth station suffers. Assumed transmission mode is 54M mode of CSMA/CA. When nine stations try to send long packets, a VoIP packet from 10th station may suffer approximately 200 ms delay.

FIGURE 27

Example of average mouth-to-ear delay



Each delay contribution to mouth-to-ear is also depicted in the Fig. 27. A 165 ms delay contribution does not depend on FWA characteristics. In case that *m* is small, this 165 ms delay becomes the dominant contribution to mouth-to-ear delay. On the other hand, as *m* increases, FWA-related contributions grow drastically. FWA-related contributions consist of:

– D(FWA): IPTD occurring in FWA portion, and

– D(Buff): Processing delay component for DV(FWA) compensation by de-jitter buffer.

And in this example, D(Buff) is comparatively larger than D(FWA).

4.3 Example calculation of R-value

As described in § 4.2, mouth-to-ear delay depends on the number of stations *m* under some conditions, and it is clear that *R*-value defined in ITU-T Recommendation G.107 depends on the mouth-to-ear delay. Therefore, it can be thought that *R*-value depends on *m*. Under the assumptions described in § 4.2, an example of relation between *m* and *R*-value is shown in case that the random packet-loss probability *Ppl* is 0, 0.1, 1.0, and 3.0%, respectively. The packet-loss dependent Effective Equipment Impairment Factor *Ie-eff* given by the Equipment Impairment Factor *Ie*, Packet-loss Robustness Factor *Bpl*, and *Ppl* is listed in Table 15.

Other parameters values required for *R*-Value calculation are the default values of ITU-T Recommendation G.107. Here, it is assumed that:

Mean One-Way Delay (*T*) = Absolute Delay (*Ta*) = Round-Trip Delay (*Tr*)/2.

Examples of calculation result are depicted in Fig. 28 (54M mode) and Fig. 29 (24M mode). When *m* is small, *R*-value degradation of each mode is nearly equal. However, as *m* increases, *R*-value of 24M mode degrades intensively.

TABLE 15

Equipment impairment factor

Coder	**PLC**	***Ie***	***Bpl***	***Ppl* (%)**	***Ie-eff***
G.711	G.711 Appendix I	0	25.1	0	0
				0.1	0.4
				1.0	3.6
				3.0	10.1
G.711	None	0	4.3	0	0
				0.1	2.2
				1.0	17.9
				3.0	39.0

In case that *Ppl* is small like 0.1%, it is found that ITU-T Recommendation G.711-compliant coder without PLC can be used for voice communications. However, under conditions that much packet loss occurs, PLC will be needed. PLC improves *R*-value degradation and keeps *R*-value over 70%.

FIGURE 28

Example calculations of *R*-value (CSMA/CA, 54M mode)

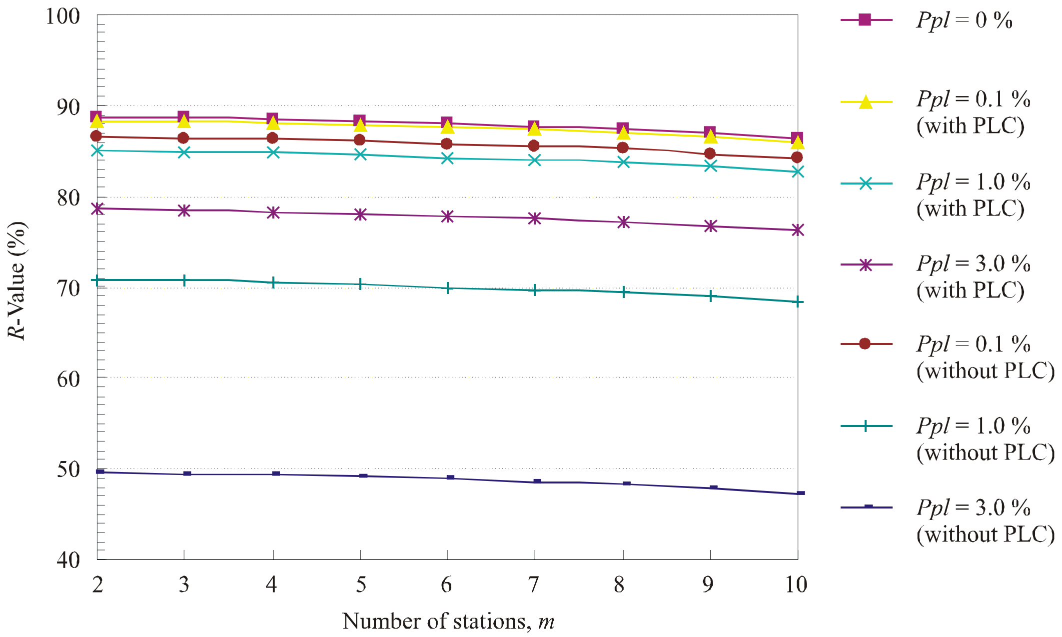


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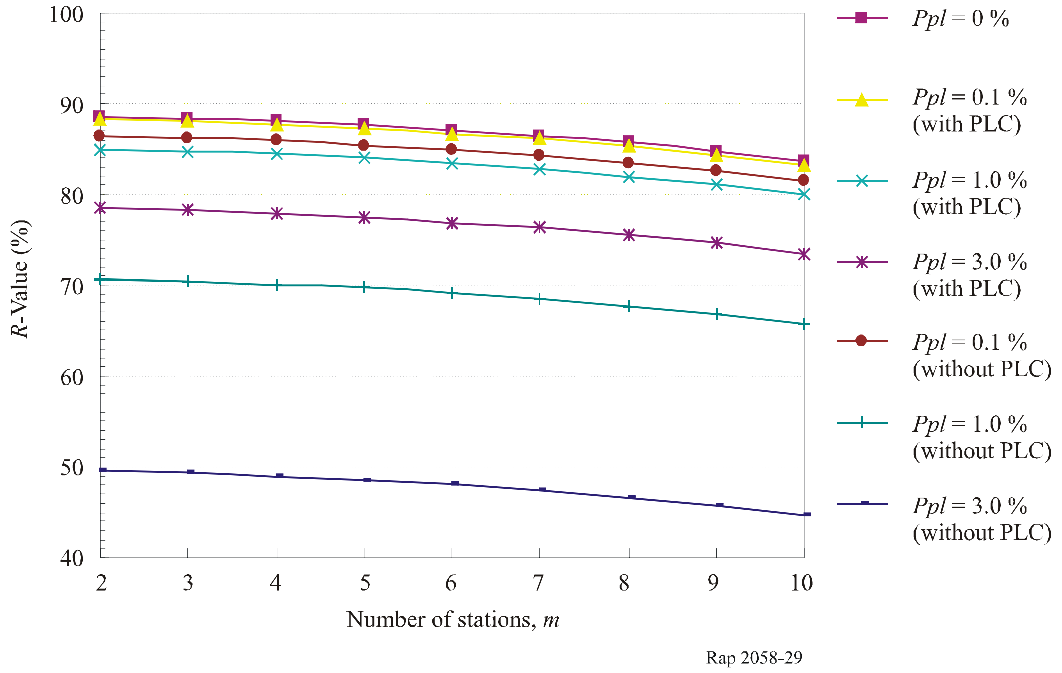
FIGURE 29

Example calculations of *R*-value (CSMA/CA, 24M mode)



Annex 6

Techniques to improve spectrum utilization efficiency

1 Introduction

Cells are generated asynchronously in ATM systems. Idle cells are inserted when the cell stream is transmitted over broadband transfer lines as shown in Fig. 30. In this case, valid cell intervals and CDV are maintained. However, this method is not suitable for wireless systems from the viewpoint of spectrum utilization efficiency. It is effective to remove the idle cells to increase spectrum utilization efficiency. Methods to keep CDV without inserting idle cells are presented in this section. They are examples of different possible approaches.

Idle cell removal is essential to guarantee spectrum utilization efficiency. Idle cells should be considered as a physical layer filling to be transmitted when nothing useful is available for transmission. Idle cell removal, anyway, does not imply that ATM cells must be transmitted in bursts longer than one cell, because transmitting ATM cells in burst longer than a single cell produces an unacceptable delay. Each time a burst forming is necessary (like on the upstream channel of a P-MP system) it is required that bursts be a single cell long.

Possible ways to avoid idle cells on the radio channel are:

– Statistical multiplexing (if radio channel bit rate is comparable with that of baseband interface).

– Assigning a much lower bit rate to the radio channel.

2 Statistical multiplexing

When statistical multiplexing (shared use of a physical medium by more than one connection/station) is present and the radio channel bit rate is comparable with (or not much slower than) the rate of baseband interface, it is possible to reduce the waste caused by idle cells without introduction of cell delay variation. A second connection/station will use the time slots not occupied by the first station and so on. Figure 30 is an example of one connection: time slots left empty after removal of idle cells (marked by "*") can be used by other connections/stations on the radio channel.

3 Bit-rate reduction over the radio channel

If radio channel bit rate is much lower than baseband interface bit rate at each terminal station, there is a granularity loss in the positional information of cells; a technique like those described in the following can be used to recover that information; this is probably required only if the reduction of bit rate is large (> 10 times) and connection "burstiness" is high (long bursts at peak cell rate (PCR) followed by long silence periods). It can also be noted that negotiated PCR should not exceed the radio channel rate (connection admission and control is responsible for that) and in this hypothesis, the problem does not arise. If one wants to allow the PCR to exceed the radio interface rate one of the following techniques can be used. They are provided as examples.

If a radio channel bit rate is much lower than baseband interface bit rate, idle cells are removed at the transmitting side but must be reinserted at the receiver. At the receiving side there are two methods to regenerate idle cells.

a) Regular interval

In this method the valid cell interval is kept fixed at the receiver side as shown in Fig. 31. This method is suitable for CBR with constant cell interval and unspecified bit rate (UBR) that operates independently of CDV. The advantage of this method is an easy control function. On the other hand, a disadvantage is the degradation in performance due to CDV in the VBR service.

b) Time stamp

In this method, timing information is generated according to each cell time slot, and this information is stamped to the cell at the transmitter side as shown in Fig. 32. Cell intervals are exactly controlled according to the time stamps at the receiver side. The merit of this method is maintenance of CDV. The demerit of this method is decrease in transmission efficiency due to additional bytes required for the transmission of time stamps and an additional delay necessary to handle CDV fluctuations (the amount of additional delay is related to the CDV fluctuations and maximum burst length on the baseband interface and therefore it is not easily derivable).

FIGURE 30

High speed wireless connection with burst one cell long and statistical multiplexing

* = Idle cells. In the wireless connection idle time slots are used for other connections (statistical multiplexing)

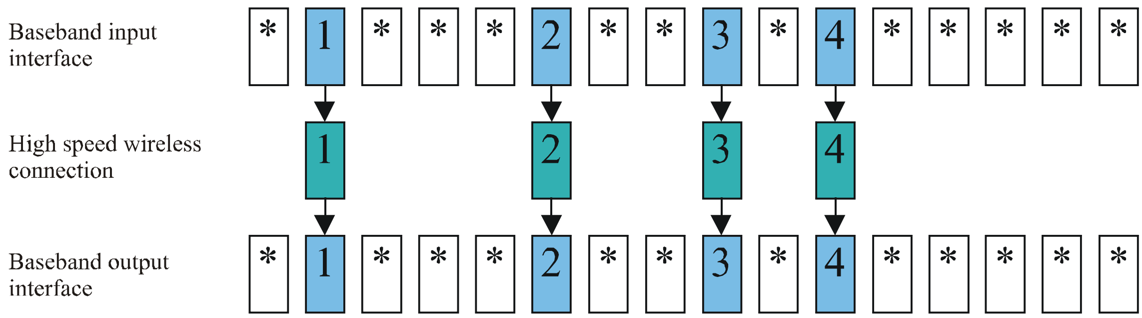


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FIGURE 31

Low speed wireless connection with burst one cell long and no time stamping (CDV not maintained)

* = Idle cells.

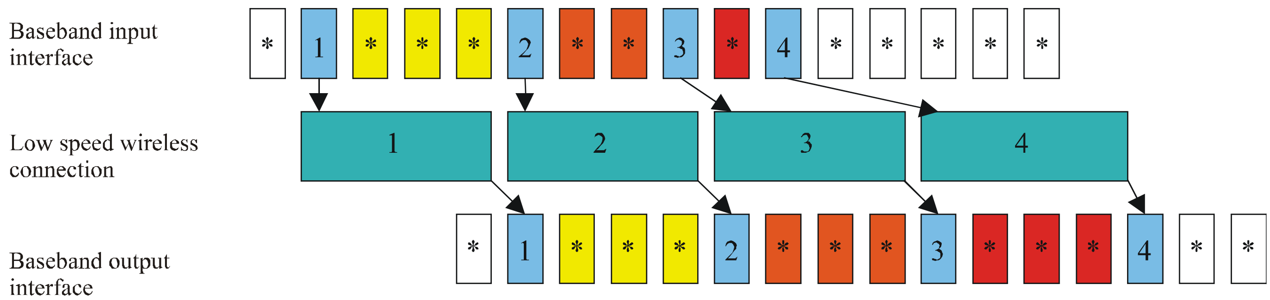

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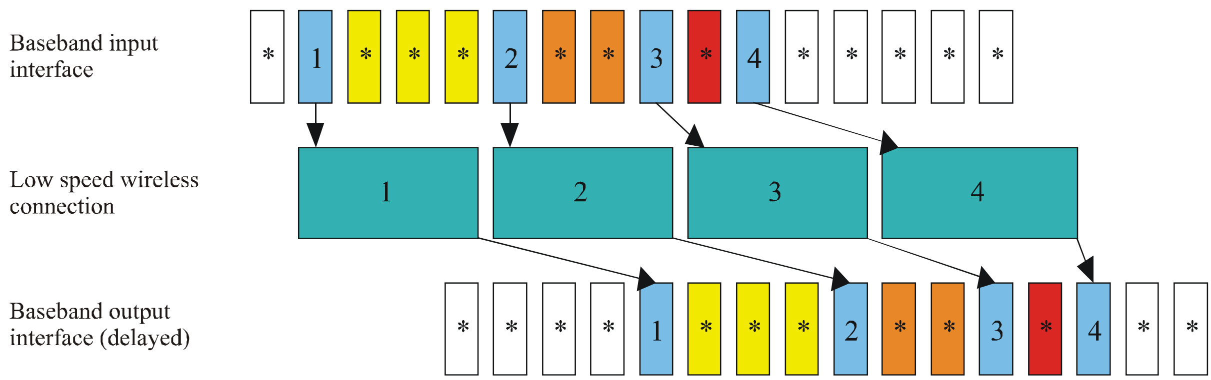
FIGURE 32

Low speed wireless connection with burst one cell long and time stamping (CDV maintained)

* = Idle cells.



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Annex 7

Error correction techniques in an ATM-based FWA system

1 ARQ for an ATM-based FWA system

1.1 Introduction

ARQ is one of the error control techniques. ARQ needs an error detection code and a sequence number for each cell. The sequence numbers are generated according to the order of cells and added to each valid cell at the transmitting side. When a valid cell is lost, the receiver requests retransmission of the lost cell to the transmitting side. After the retransmission, cells are rearranged according to the sequence number.

Such ARQ is effective for non-real-time services. However, delay time becomes a problem in real-time services. One solution is to limit ARQ only to non-real-time services. However, a fast ARQ strategy with limited number of retries can be applied to real-time services.

A comparison of ARQ versus FEC only or a combination of the two techniques should be evaluated.

The major drawback of ARQ with ATM is delay; to avoid CDV a fixed delay equal to maximum retransmission time (or time to retransmit the maximum number times) must be added at the receiver.

In an ATM system there are usually services with strict delay requirements and services less sensitive to delay; cell loss requirements must also be taken into account.

ATM Forum wireless ATM working group suggests the following as strictest requirements[1]:

	Real-time services	Non-real-time services
CLR	10^{-7}	10^{-9}
Delay	10 ms	500 ms or higher

Two options are available: define a fast ARQ strategy on all the traffic or introduce ARQ only for non-real-time services.

The choice to restrict ARQ use only to non-real-time services implies that the radio channel must be dimensioned (transmitted power and FEC capacity) to guarantee a cell loss ratio lower than 10^{-7} without ARQ; the use of ARQ for non-real-time services allows to reduce CLR of non-real-time services to the required 10^{-9} which corresponds to a 0.5-1 dB gain (over a Gaussian channel; therefore it only gives marginal advantage (the same gain can be obtained with negligible increase of power or FEC length) and, moreover, it is complicated.

The restricted ARQ could also lead to situations in which the system will be out of service for CBR and will be operational for UBR. This is a major reason why it is questionable whether a retransmission which is slow and selective over the type of service is useful.

It is recommended that a fast selective repeat ARQ with potential retransmission of all types of traffic is considered as optional especially to cope with channels affected by burst errors (those

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[1] ATM Forum Baseline Text for Wireless ATM specification, February, 1998 (ATM Forum BTD-WATM-01.06).

dominated by fading and interference); a maximum number of retransmissions and a maximum retransmission delay (which can in principle be different connection by connection and can fall to 0 = no retransmission) will allow dealing with connections with different requirements.

All possible care must be taken to make retransmission as fast as possible in order to allow a larger number of connections to be treated by ARQ.

It can be shown that, if a channel is Gaussian, the ARQ advantage is limited compared to FEC with the same overhead (or a longer FEC if used in conjunction). ARQ is powerful in a burst error environment. Lower frequencies (in particular for mobile but also for fixed access) are subject to man-made noise, interference and multi-path fading and burst errors are quite common at these frequencies. Higher frequencies are less subject to the above-mentioned types of noise and therefore they are typically considered white Gaussian noise environments.

ARQ should be recommended (optionally) for lower frequencies but not for higher (over 18 GHz).

Special care must be taken to implement ARQ in TDD systems (both in P-MP and P-P); because it causes high delays (upstream and downstream frames longer than one ATM cell are foreseen and therefore an acknowledge and a retransmission are always delayed by a frame time). It is recommended that ARQ in TDD systems only be considered where necessary.

1.2 Comparison on different ARQ approaches

In general, there are two methods for ARQ, i.e. go-back-*n* method (GBN) and selective repeat method (SR). The schemes of GBN and SR are shown in Fig. 33 a) and b), respectively. In GBN, the receiver gives the transmitter the sequence number of the first error cell as a NAK, then the transmitter re-sends the cell stream from the cell indicated by NAK. In SR, the receiver, upon reception of a NAK, will retransmit only the error cell.

In addition to these two methods, there could be the method by which the transmitter receives confirmation of successfully received cells from the receiver one by one, and does not transmit the subsequent cells until the prior cells have reached the receiver (see Fig. 33 c)). This cannot be acceptable for FWA systems using ATM from the viewpoint of frequency utilization.

SR needs more complicated ARQ control than GBN, but has higher efficiency. Figure 34 shows the characteristics of transmission efficiency of GBN in case of some TDMA-based frame format, assuming random error. The throughput of SR equals to that of GBN when $N_{output} = 1$, and is independent on N_{output}, where N_{output} indicates the number of cells transmitted during ARQ period (i.e. TDMA frame). In this calculation, there is no limitation to the number of repetitions. An unlimited buffer capacity at the transmitter is assumed, and the transmission volume of ACK information is ignored for simplicity. If burst errors tends to occur rather than random errors, however, the difference between these two methods becomes smaller. As shown in Fig. 34, on the condition that the BER is high, the transmission characteristics of GBN are remarkably degraded compared with those of SR. Consequently, SR is desirable for ARQ.

1.3 Methods to send error notifications from the receiver to the transmitter

It is common in SR that the receiver informs the transmitter with ACK of the sequence numbers of data (in this case, cells), which have reached the receiver. The ACK information for all transmitted cells, however, may be huge in the wireless ATM systems discussed in this document, because the number of cells managed by ARQ is large. In this case, the ACK information always occupies a large part of the channel bandwidth. Therefore, the method by which the receiver informs the transmitter of NAKs indicating cells not yet received by the receiver is desirable from the viewpoint of frequency efficiency (under the assumption that the retransmission incidence is lower than 1% so that a NAK is only transmitted every 100 cells).

FIGURE 33

Basic ARQ schemes

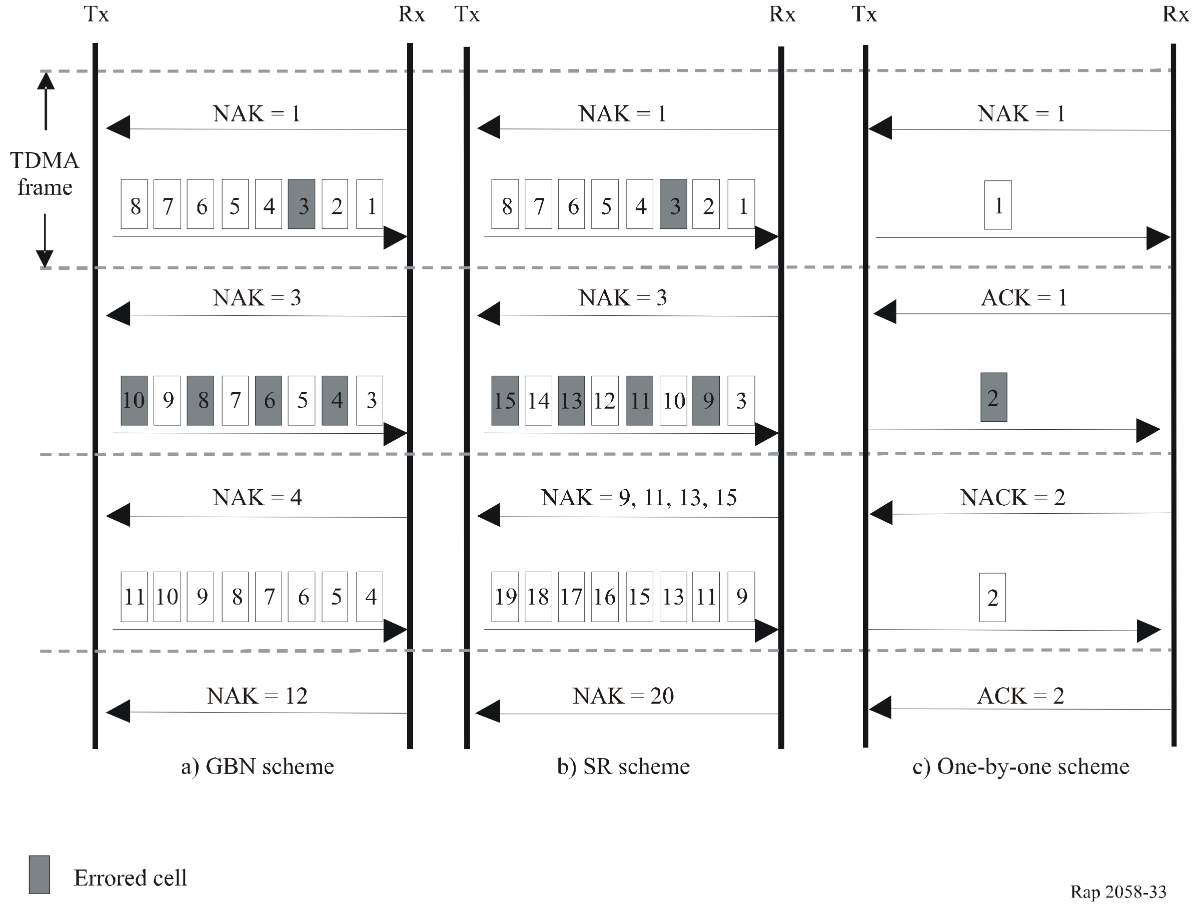

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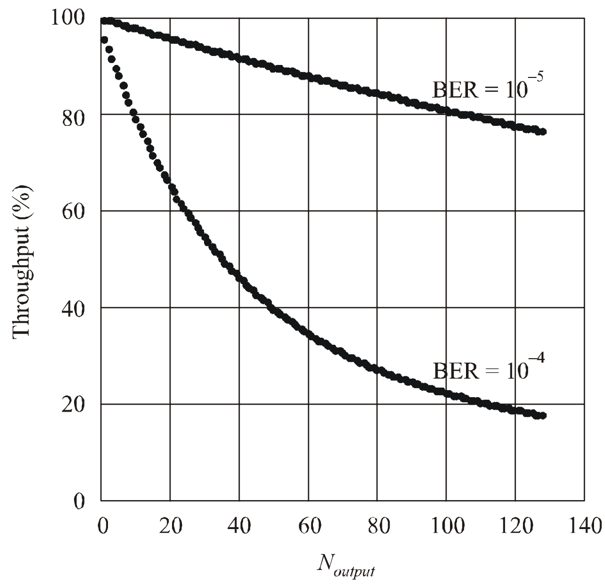
FIGURE 34

Relationship between N_{output} and throughput of GBN [2]



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[2] OHTA, A. *et al*. [May 1998] PRIME ARQ: A Novel ARQ Scheme for High-speed Wireless ATM – Design, implementation and performance evaluation, VTC'98, p. 1128-1134.

The use of NAKs (without ACKs) to reduce acknowledge traffic requires more details: the NAK for cell 2 can be issued only after reception of cell 3; if the NAK is lost or the first retransmission is lost, the NAK must be repeated by the receiver after a time-out. This timeout can be short if NAKs and retransmissions are assigned the highest priority.

Another problem that must be addressed when the choice to use only NAKs is made is the following: the transmitter must keep the cell in a buffer until it can be sure that a NAK will not be received. Unfortunately, considering the possibility of NAKs being lost, this means that the cell must be retained until a (long) timeout expires. This timeout must be longer that the maximum retransmission time or the time required for the maximum number of retransmissions. The same buffer (and therefore a fixed delay) must be added in the receiver to allow reconstruction without introduction of CDV.

Notwithstanding the above, the problem of acknowledge traffic especially in P-MP upstream direction is very serious though and NAKs are a good idea to cope with that even if it has some disadvantages. Another way is to adopt a cumulative acknowledge technique. This technique causes an intermediate performance between go-back-*n* and selective repeat but combines low acknowledge traffic advantage with smaller buffers.

1.4 Avoidance of excessive repetition

In wireless ATM, several kinds of services are dealt with, and each service has different QoS requirements. The required quality of real-time services such as CBR and rt-VBR (real-time VBR) in wireless ATM systems is being specified at some standard organizations. In the ATM Forum the required values of CTD and CLR in the most severe case of CBR or rt-VBR are considered to be 10 ms and 10^{-7}, respectively. In addition, according to ITU-T Recommendation I.356, CLR should be less than 10^{-8} for the most severe class. In order to enhance adaptation to the area of FWA, it is desirable that CLR can meet the above qualities in higher error wireless environments by error correction using FEC and/or ARQ.

In real-time services the allowable delay time is small as mentioned above. The excessive trial of ARQ over the allowable delay time is no longer meaningful. Therefore the receiver should watch the number of repetitions or delay time, and when one of the two exceeds the allowed value the receiver should stop issuing NAKs (the transmitter will also discard the cell from its buffer after the same timeout).

2 Error correction in an ATM cell transmission

2.1 General considerations

In FWA conveying ATM cells, objectives in ATM layer performance should be met as well as those in physical layer performance. Relations between the performance parameters in both layers are now being studied under Question ITU-R 210/9. In particular a system must be designed to satisfy the SESR objective in ITU-T Recommendation G.826 and also meet the CLR or CER objectives specified in ITU-T Recommendation I.356.

2.2 HEC vs. FEC and effects of differential encoding

In ATM cell transport HEC is normally applied to the header portion. Therefore, one bit error in a header portion can be corrected, resulting in much fewer lost cell outcomes (LCOs) and misinserted cells.

However, it should be noted that HEC is not effective when FWA employs differential coding in which one code error causes two contiguous bit errors. This may cause HEC to become ineffective and thus affect CLR and CMR performance of the system. It should also be noted that a HEC is probably not powerful enough for most wireless applications and a more powerful FEC should be added to the whole ATM cell.

FEC is applied as a correction method to the whole ATM cell and it is usually capable of correcting multiple errors. If it is applied, it solves the problem of differential encoding. Moreover if a FEC is introduced, the HEC is redundant and can be removed. It can be easily verified that a one-byte-longer FEC is always more powerful than a FEC+HEC combination.

This suggests that FEC be applied and HEC be removed whenever applicable. If FEC is not applied and differential encoding is used a method is described in the following to make HEC work in a differentially encoded environment.

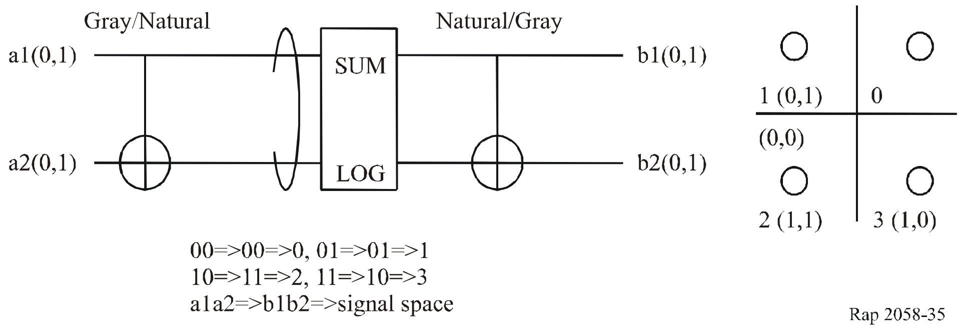
2.3 Example of a method to avoid effects of differential coding when HEC without FEC is applied

Although there are digital paths for which differential coding is not necessary, it is in many other cases, of differential coding. In order to avoid the error propagation within a single cell, it will be necessary to adopt some kind of bit signal processing methods among the input ATM cell streams. The following provides an example of this method.

Differential coding is generally used in order to eliminate the uncertainty of the phase of the recovered carrier. In this system, the modulation signal is mapped on the signal space according to the sum of two contiguous signals. In the receiver side, the difference between the receiving signal and the previous one is calculated. Summation and differentiation is calculated using modulo-4 logic algorism. Even if there is an uncertainty of 90º, this is cancelled because the recovered carriers of the two adjacent signals have the same phase. These relationships are shown in Fig. 35.

FIGURE 35

Differential coding



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However, if the differential coding is used, a bit error occurrence in the receiving signal may propagate to two consecutive time slots as shown in Fig. 36.

FIGURE 36

Consecutive error due to the differential coding

b1 OOOXOOO=>a1 OOOXOOO or OOOXXOO or OOOOXOO or OOOOOOO
b2 OOOOOOO=>a2 OOOOXOO or OOOOOOO or OOOXOOO or OOOXXOO

X: error, O: no error

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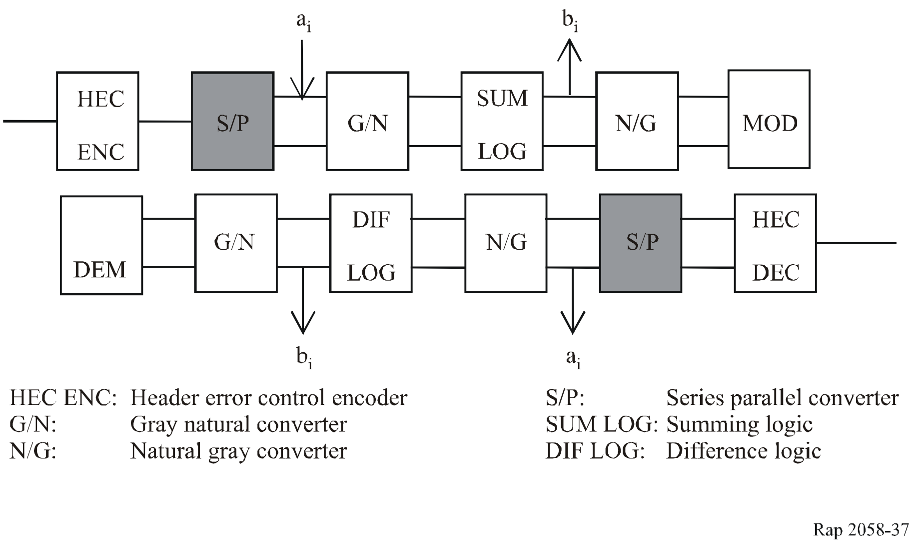
On the other hand, HEC control is exercised in the ATM cell header. The HEC is processed outside the differential coding process as in Fig. 37, and it is influenced by the above consecutive error due to the differential coding. Since HEC employs such function of error correction for one-bit error and error detection for more than one bit errors that the error correction effect will be remarkably deteriorated if the above consecutive errors due to the differential coding.

FIGURE 37

HEC process of ATM cell



In order to avoid the HEC deterioration, bit-interleaving is effective as shown in Fig. 38. The bit-interleaving process breaks up the consecutive bits to the header and the payload inside the cell. This will break up the consecutive error bits to the header and the payload, resulting in one-bit error in the header and improving the probability of successful error corrections. In Fig. 38, the cell is divided into four, assuming differential encoded QPSK.

FIGURE 38

Bit-interleaving inside ATM cell