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Medium access for narrowband wireless ad-hoc networks; requirements and initial approaches

Lars Erling Bråten, Mariann Hauge, Jan Erik Voldhaug and Knut Øvsthus

Forsvarets forskningsinstitutt/Norwegian Defence Research Establishment (FFI)

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Approved by

Vivianne Jodalen	Project manager
Torleiv Maseng	Director of Research
Vidar S. Andersen	Director

English summary

The initial requirements for a tactical military mobile ad-hoc network are discussed and implications on the design of the link layer for a narrowband system have been investigated. The main focus is on medium access control (MAC) protocols suitable for time division multiple access (TDMA) based mobile VHF and UHF combat networks carrying both data and voice traffic. The work is carried out as part of a NATO SC6 attempt to define a CNR network enabling standardised communications between nations employed in joint operations.

In this initial time slotted design the narrowband system is assumed to occupy 25 kHz of spectrum, thus efficient transfer of information and reasonably low overhead ensuring scalability is required to support networks of various sizes. Support for quality of service classes, prioritisation and pre-emption is required. The access to the shared radio channel is managed by the link layer MAC protocol. Terrain obstacles, interference, jamming and potentially long distances may lead to multiple hops, and the distributed resource allocation should handle both hidden and exposed nodes in a time dynamic network topology. The MAC protocols should, in cooperation with the network layer, offer uni- and multicast as well as for example regular transmission of position updates of nearby friendly forces.

A literature review has been performed and interesting MAC concepts identified. Furthermore, network timing approaches are studied to some extent. Network splitting and merging as well as radio based combat identification (RBCI) are examples of topics briefly discussed.

Dynamic time division multiple access (D-TDMA) and soft reservation schemes such as collision avoidance time allocation (CATA) are identified as potential solutions fulfilling most of the above mentioned requirements. The main challenge for both approaches is to limit overhead due to signalling of control messages while at the same time fulfilling the requirements. We have developed an initial link layer design for two candidate approaches. Initial performance assessments are presented and the characteristics of the two alternatives are compared. For a slowly varying network topology and relatively long traffic flows, as experienced for combat networks operating at VHF frequencies, D-TDMA seems to be the most promising approach with respect to the available traffic capacity. With increasing node mobility, higher operating frequency or a more bursty traffic pattern, the soft reservation approaches may become viable alternatives. In such cases reservation signalling would for example occupy more of the available radio resource when utilising D-TDMA. Examples of possible signalling approaches are given for both uni- and multicasted traffic over one or more hops. For both approaches instability of the contention mechanism(s) during heavy load requires a form of connection admission control to ensure successful outcome of the process for start-up of new traffic flows.

Sammendrag

I forbindelse med standardiseringsarbeid innen NATO SC6 for et fremtidig militært mobilt ad-hoc nett har vi sett på kravene som kan stilles til de ulike protokollagene. Vi har tatt utgangspunkt i et forslag til fysisk lag (modulasjon og koding) utviklet av Communications Research Centre Canada (CRC). Videre har vi utført en forstudie med fokus på hvordan linklaget kan realiseres for et moderne VHF/UHF radionett med både tale og datatjenester innenfor en begrenset båndbredde på 25 kHz. Hovedutfordringen har vært å identifisere og videreutvikle medium aksess kontroll (MAC) metoder som egner seg for militære nett som benytter tidsdelt multippel aksess. MAC protokollen styrer adgangen til en felles radiokanal delt av flere av nodene i nettet.

Med lav båndbredde øker kravet til effektiv overføring av informasjon. I tillegg bør den valgte løsningen være skalerbar med hensyn på antall noder (radioer) i nettet, da bruken kan variere fra for eksempel å dekke deler av en brigades kommandonett til mindre interoperabilitetspunkter mellom ulike nasjoner under felles operasjoner. Støtte for forskjellige klasser av tjenestekvalitet, militær prioritet og mulighet for at høyprioritets trafikk kan koble ned lavprioritetssamband og ta over kapasiteten til eget bruk (*preemption*) er noen av kravene som stilles. Terrenghindringer, interferens, jamming og lange avstander kan kreve flere hopp mellom sender og mottaker(e), og distribuert ressurstildeling som håndterer skjulte og eksponerte noder i et nett med tidsvarierende topologi er nødvendig. I samarbeid med andre protokollag skal MAC protokollen kunne tilby både uni- og multicast tjenester med varierende krav for pålitelig levering. Både sanntidstjenester som for eksempel tale og ikke-sanntidstjenester som for eksempel e-post skal håndteres.

Vi har utført en litteraturstudie for å finne egnede MAC protokoller som støtter de definerte kravene. Videre har vi studert tidssynkronisering i nettet og diskutert hvordan nye noder kan koble seg til nettet, samt hvordan deling og sammenslåing av nett kan tenkes utført.

Dynamisk tidsdelt multippel aksess (TDMA) og ”myk” reservasjon peker seg ut som to aktuelle kandidater for videre undersøkelser. Hovedutfordringen for begge metodene er å begrense mengden kontrollinformasjon for å sikre god overføringseffektivitet samtidig som tidsvariasjonen i nettet håndteres. Utvikling av designeksempler for mulige implementeringer ga mulighet for å sammenligne ytelsen og egenskapene til de to MAC konseptene. Dynamisk TDMA synes velegnet for mobile ad-hoc nett der topologien varierer relativt sakte og trafikkflytene har relativt lang varighet. Dette er typiske egenskaper for VHF kommandonett som opererer innen 30 – 88 MHz. Med økt nodemobilitet, økt frekvens eller høyere andel av skur trafikk vil protokoller som benytter myk reservasjon bli en aktuell erstatning for dynamisk TDMA. Dette vil typisk kunne inntreffe for ad-hoc nett som benytter militære UHF frekvenser i for eksempel området 225 – 400 MHz der topologidynamikken varierer raskere enn ved lavere frekvenser. Videre har vi beskrevet eksempler på mulige signaleringsmetoder for både uni- og multicast for de to protokolltypene både for ett-hopps- og multihopp kommunikasjon. Videre er god håndtering av trafikkøer påkrevd når lasten i nettet er høy, slik at nye forbindelser ikke kobles opp dersom det ikke er kapasitet i nettet.

Contents

Preface

This report describes a study of MAC-protocols carried out in connection with ongoing work at FFI to support the development of interoperable mobile ad-hoc networks within NATO. The main goal is efficient wireless interworking and thereby communications between different national forces during multi-national operations. An additional goal is to obtain a system enabling communication network based operations, popularly denoted as network enabled capabilities (NEC).

1 Introduction

The NATO ad-hoc working group on VHF and UHF systems has started development of narrow- and wideband ad-hoc wireless networks operating at VHF and UHF frequencies. Mobile ad-hoc networks (MANETs) consist of a group of wireless terminals that dynamically form a multihop network. The intention is that the NATO standardised communication system will enable scalability both with respect to bandwidth, operating frequency and number of terminals. The initially targeted users for the narrowband system are army platoons with typically 10 to 30 (less then 200) radios in a potentially hostile jamming scenario. It is reasonable to assume that the wireless system should scale reasonably well when increasing the number of users, and features such as multi frequency operations might be of interest. The main service requirement is offering voice and low rate data communication in the form of position information, messages and background transfer of files such as maps and photos. Connectivity outside the ad-hoc network is expected to take place from one or more gateways located at a command post utilising an overlaying hierarchical high capacity wireless network such as terrestrial or satellite communications. The network should be established ad hoc, that is, without relying on existing infrastructure.

Communications Research Centre Canada (CRC) has started to develop a near constant envelope modulation time division multiple access (TDMA) approach combined with iterative forward error correcting coding for the physical layer. This approach is intended for both fixed frequency and in frequency hopping mode [1] [2]. Telefunken has started developing a higher capacity linear quadrature amplitude modulation (QAM) modulation with low peak to average power ratio. The interface between the physical layer and higher layers, such as medium access control (MAC), has been defined as well, enabling cross layer approaches. Future work may include linear modulation utilising efficient iterative decoding techniques such as Turbo coding or low density parity check (LDPC) coded QAM modulation.

Link layer (air interface) encryption is currently an open issue, so is radio resource management and other required link layer functionality. The objective is to provide network layer interoperability utilising IPv4/IPv6. Radios capable of frequency hopping will utilise a stable time source, fixed frequency radios may require timing establishment from the network itself, utilising lower cost time sources in addition to information from the physical layer.

Terrain obstacles, interference, jamming and long distances may lead to several hops between the originating node and the destination node(s). Portable devices with limited transmit power and sensitivity will have shorter range than for example vehicle mounted radios with external antenna and high performance transmitter and receiver. The operational frequency and node mobility will influence the connectivity as well, where for example at UHF frequencies a node may require several hops to reach the destination while at VHF a single hop is sufficient.

The access to the radio channel shared within the ad-hoc network is managed by the MAC protocol which is part of the link layer. MAC protocols can be divided into three main classes according to [3]: contention, allocation and hybrid protocols combining contention and allocation. The communications within the ad-hoc network is normally of peer-to-peer type without any central resource management node. Thus, the network management in ad-hoc network is distributed between the participating nodes forming a self-organising network. In this study we focus on MAC protocols for combat net radios (CNR) carrying voice and data communications. The main requirements are discussed in Section 2. Previously reported works are discussed in Section 3 followed by a pre selection of promising MAC algorithms for further simulation studies. Initial link layer design considerations are discussed in Section 4. Finally the conclusions from the current work are offered.

2 Requirements

NATO has started to work on the general requirements for a narrowband waveform, see [4]. Figure 2.1 illustrates a multinational brigade where basic communications flows are shown, typically vertical for command and control and horizontal for direct sharing of situational awareness.

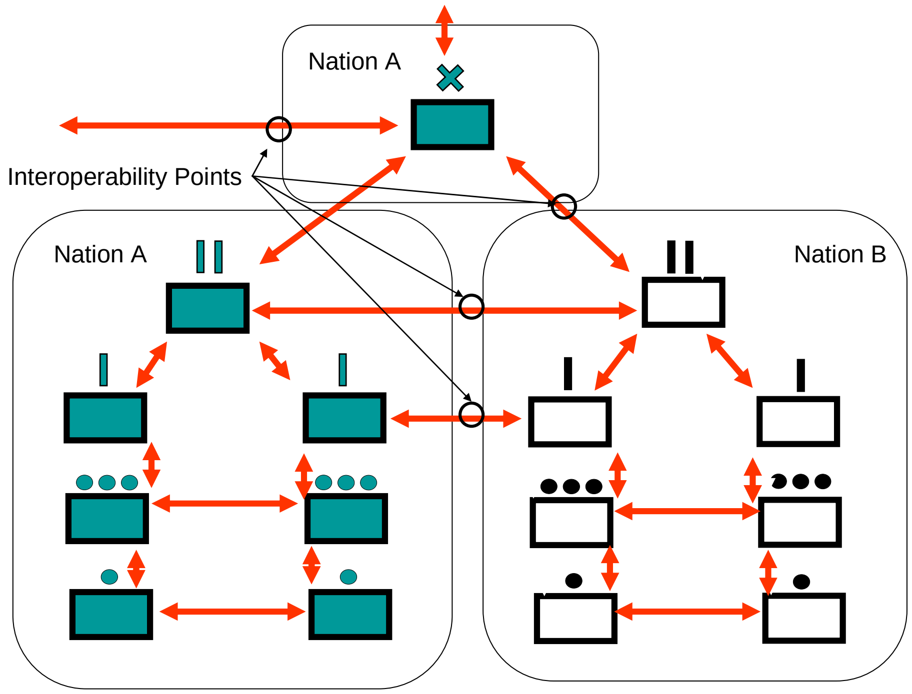


Figure 2.1 Information flows and interoperability points, adopted from [4]

One of the current communication challenges is to obtain efficient horizontal communications between different national forces at the same level in the organisation.

The main objective of this section is to identify system requirements relevant for the link and network layers. However, the ability of the physical layer to handle different terrain types, frequency ranges and transmission capacities is of interest as well. The different layers of a possible future standardisation agreement (STANAG) are depicted in Figure 2.2

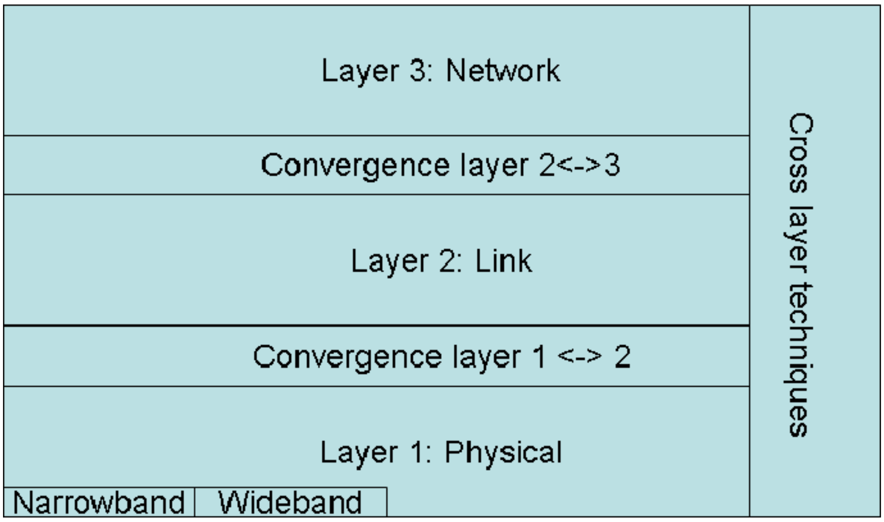


Figure 2.2 VHF-UHF possible STANAG collection for the MANET

The tasks normally assigned to the link layer include framing and segmentation, medium access control and assignment of physical layer burst mode (modulation, coding, timing etc). In addition retransmission schemes may be implemented at the link layer. The physical layer implements the over the air transmission, including modulation, coding and features such as frequency hopping. In between the physical layer and the link layer, the air interface/link layer encryption, protecting both user traffic and system signalling from traffic analyses, may reside. The network layer and layers above provide end-to-end routing (unicast, multicast and possibly anycast) and different end-to-end data-transport services to the end-user applications.

The objective of this section is to provide initial requirements for the link and network layers based on the applications and services considered important for a tactical mobile wireless system. It is expected that cross layer approaches will be utilised to improve the performance. Convergence and interoperability with other communication systems will take place at the network layer with Internet protocols for both transport of user data (IPv4/6), end-to-end security soulutions, and required network management functions handling for example routing and QoS.

2.1 User requirements

The targeted users are military forces with less then 200 radios (typically 10 to 30) in a potentially hostile jamming scenario. Several adjacent networks (domains) should be able to interact efficiently. The main requirement is half duplex voice and low rate data communication in the form of position tracking, messages and background data transfer. Full duplex voice may be required for interworking with fixed or mobile telephony systems. It is not required during normal operation of the combat net and is not considered further in this study.

The military users are expected to utilise one-to-one data and voice communications (unicast) as well as multicasting within one or several groups of users. Voice communications utilising push-to-talk will be an important application within groups of different sizes. Voice communications should be of good quality even when background noise is present, and time delays should not degrade the quality experienced by the users (QoE). Some details regarding voice encoding and decoding is given Section 2.2.1. Quality of service (QoS) features enabling prioritisation of traffic, users and perhaps terminals will be necessary. Pre-emption enabling high priority traffic to suspend or terminate lower priority traffic should be included due to special military operational requirements. Additional quality of service requirements are given in Section 2.3.1.

User friendly man-machine interfaces and ease of use are important factors. In this work we will not work on this directly. However, there may be implications of user friendliness relevant for the link and network layers including auto configuration, predefined user groups etc.

The users expect that information and network security aspects such as authentication of users, terminals and networks as well as privacy (encryption) are implemented. It is understood that some of these aspects are outside the scope of the current work; one example is end-to-end encryption. Radio interface (link layer) encryption to hinder traffic flow analysis is clearly within the scope of a possible STANAG, however, it is not the main focus of the current work. Reliability of the equipment in hostile physical and radio environments are important to obtain users trusting the services offered by the radio system. The availability of services in terms of insignificant outage time has implications on the network design and may well require the features of ad-hoc networks enabling multi-hop communications. The network should be able to operate in all types of terrain environments and indoor to outdoor operations may be important.

The wireless network should be able to operate over a reasonable time without any logistic support. This has implications on for example battery drain and network time synchronisation. Single users or groups of users should be able to bring their terminals into radio silence. It would be beneficial if they where still able to receive traffic during radio silence.

Interoperability with ships and airborne platforms is expected to be of significant importance. The handling of mobility is thereby a prerequisite, possibly including both terminal mobility and user mobility. Seamless handover between networks is currently not an issue. It is assumed that the wireless system is able to transport Internet protocol (IP) traffic and that it has standard interfaces, for example Ethernet or Bluetooth, enabling connection of various end-user devices.

2.2 Application and services

The main objective of this section is to identify typical services and their characteristics and requirements. Most wireless systems define service classes for e.g. real time and best effort services, enabling design of traffic bearers tailored to the services. There have been some efforts in NATO to define common architectural elements for terrestrial wireless systems, see [4]. The NC3A report presents the operational view for wireless communications in the land tactical domain. The document is being used to form the basis of the systems view, currently in development. Both of these views will be used to clarify and refine the operational and system requirements for narrowband and wideband waveforms (NBWF and WBWF).

The types of information to be exchanged and services used were in [4] summarised as shown in Table 2.1.

Type	Characteristics
Voice/Video	Real-time, some resilience to errors, unpredictable traffic patterns, group communication dominate over one-to-one communications
Position tracking	Small, regular messages with some time-relevance. This may include situational awareness functional services for combat forces.
Targeting	Timeliness, authenticity, robustness
Core services	E-mail, photographs, maps etc. Higher data quantity, lower timeliness requirements.
Functional services	Functional services for non-combat forces. This may include database access or replication. Timeliness and data quantities will be dependent on the specific service being utilised. These may be significant quantities of data e.g. database replication or timely information but usually not both.

Table 2.1 Information exchange services, adopted from [4]

The range of services requested clearly requires a wide range of possible QoS configurations. In the following some of the services are described in more detail.

2.2.1 Voice

Half duplex voice is expected to be an important application, with push-to-talk group functionality. It is expected that voice conversations normally are multicasted within groups, resembling the broadcast nature users are accustomed to. The MELPe (Enhanced Mixed-Excitation Linear Predictive) vocoder [6], [7] is the narrowband voice codec that is expected to be utilised in the ad-hoc network. It is known as STANAG 4591 [8], and operates at bit rates of 600, 1200, and 2400 bit/s. MELPe is based on the MELP voice coding algorithm described in the original MIL-STD-3005 [9]. Added features in MELPe include improved noise filtering, several encoding rates (MELP was originally only defined for 2400 bit/s), and enabled transcoding of compressed bit stream.

At 2400 bit/s MELPe samples the voice signal at 8000 Hz and divides it into packets of 180 samples, equalling 22.5 ms. In addition the encoder uses a look ahead of 161 samples, making the algorithmic delay 42.625 ms or 341 samples. Each frame is then analyzed and compressed to 54 bits. The 54 bits in a frame are mainly made up of linear prediction coefficients, Fourier magnitudes, gain, pitch, and bandpass voicing. Forward error correction (FEC) is implemented, but only in unvoiced mode. For unvoiced frames Fourier magnitudes and band pass voicing information are not sent, and instead replaced by 13 Hamming code parity bits. For MELPe 1200 three high rate frames are grouped for quantization, leading to an increase in the output frame interval to 67.5 ms. This, together with an increased look ahead of 290 samples, gives an algorithmic delay of 103.75 ms. It is believed that the bit rate out of the voice coder has constant bit rate, possibly with a variable amount of FEC overhead to cater for periods with less voice information to be transferred.

Substantial testing of candidates submitted by NATO member nations was carried out in three phases before choosing STANAG 4591. Tests include MOS-tests, intelligibility tests, tests of different speakers, different noise environments etc [10], [11]. The effects of transmission artefacts (packet loss, delay etc.) on voice quality were carried out in phase three. Phase three was carried out in US labs only and the results are not available for this study. Phase two included one bit error case, where the encoded bit streams were exposed to random bit errors at 1 per cent bit error rate.

Investigations of the sensitivity of 2400 bps MELP to packet loss using the Gilbert loss model and appropriate error concealment techniques are reported in [12]. No subjective tests were carried out, but calculations of spectral distortion were used as quality measure. They found that MELP is robust to high bit loss and burst loss rates and report ”transparent distortion” even at 20 per cent packet loss rate. Perceptual Evaluation of Speech Quality (PESQ) is an objective measure for narrowband speech quality [13]. Scores above 3 (equals fair quality) for loss rates below 20 per cent when adding only basic error concealment (frame repetition) is reported in [14]. ITU-T recommends that total end-to-end delay should be kept less than 400 ms [15]. It also states that by limiting the end-to-end delay to 150 ms transparent interactivity can be ensured.

2.2.2 Position tracking

Tracking of the position of friendly forces is expected to be the most important data service (in addition to a text messaging/chat service) in the mobile tactical network. This service type will require an efficient broadcast/multicast distribution mechanism. Position tracking may not require a reliable transport, but need timely packet delivery. In some cases one might want full broadcast of this information, possibly requiring multihop multicasting. A transport service where old packets are dropped and the most recent packet in the queue is transmitted would be beneficial for this application type.

2.2.3 Targeting

Targeting services require low delays, authentication and robustness to errors.

2.2.4 Core services

The core services include transfer of e-mail, photographs, maps etc. Relative high data quantities may be transmitted, accepting transmission delays but not necessarily any errors depending on the application utilised.

2.2.5 Functional services

The functional services are ment to be utilised by non-combat forces. These services may include database access or replication. Timeliness and data quantities will be dependent on the specific service being utilised. These may be significant quantities of data e.g. database replication or timely information but usually not both. These services will have a low priority in the network.

2.2.6 Radio based combat identification

The objective of Radio Based Combat Identification (RBCI) is to hinder fratricide experienced in air to ground engagements, potentially it may be utilised in ground to ground situations as well. It is expected that the delay requirements on RBCI is rather strict due to low flying airplanes, and an initial assumption is establishment within about one second.

2.3 Network layer

In this section the main functionality of the network layer is initially described.

2.3.1 Quality of service

A mixture of different traffic types is likely to be admitted to the network. Different traffic types (e.g., routing, voice, position information, messages and background data transfer require the support for different classes of service from the data network. The QoS management in each node should treat each class of service differently according to its tolerance to for example time delay, jitter and loss [16]. In addition a number of other factors may be considered as QoS requests, including:

- Priority (precedence: routine, immediate, priority, flash, flash-override)
- Survivability (reliable, unreliable)
- Pre-emption
- Capacity (bandwith)

The NATO document [5] identified a compact number of quality of service (QoS) classes with accompanying time delay limits, see Table 2.2.

QoS	**Characteristics**	**Time limit**
1	Real-time	< 250 ms
2	Non-real-time, but time-critical	250 ms – 10 s
3	Non-real-time. Lower priority.	10 s – 1 minute
4	Best effort	> 1 minute

Table 2.2 Information exchange timeliness, adopted from [5]

Most of the time delay requirements are not strict, and compromises between different characteristics will have to be made during the design process.

In tactical networks the QoS architecture must handle both different classes of service, and flow precedence (e.g., Multi-level Precedence and Pre-emption (MLPP) mechanisms [17] used in line switched military communication). The network should also be able to guarantee the availability of resources for a few very important data flows. Absolute guarantees are difficult to provide in mobile wireless networks where the network capacity vary with channel conditions and network topology, but guarantees might be given with a specified probability.

Tactical networks are often divided in security domains with different classifications (including the unprotected classification). Very restricted communication is allowed between security domains with different classifications. The only communication that is currently allowed from a high classification to a lower classification in most NATO countries, is the Type of Service/Traffic Class field in the IPv4/IPv6 header. The link and network layer in the V/UHF network is therefore likely to forward packets and flows based on the QoS description in this field. This suggests that a DiffServ class of QoS mechanisms might be used.

The Interoperable Networks for Secure Communications (INSC) project [18] has shown that their suggested DiffServ classification, see Figure 2.3, has been efficient also for mobile wireless networks.

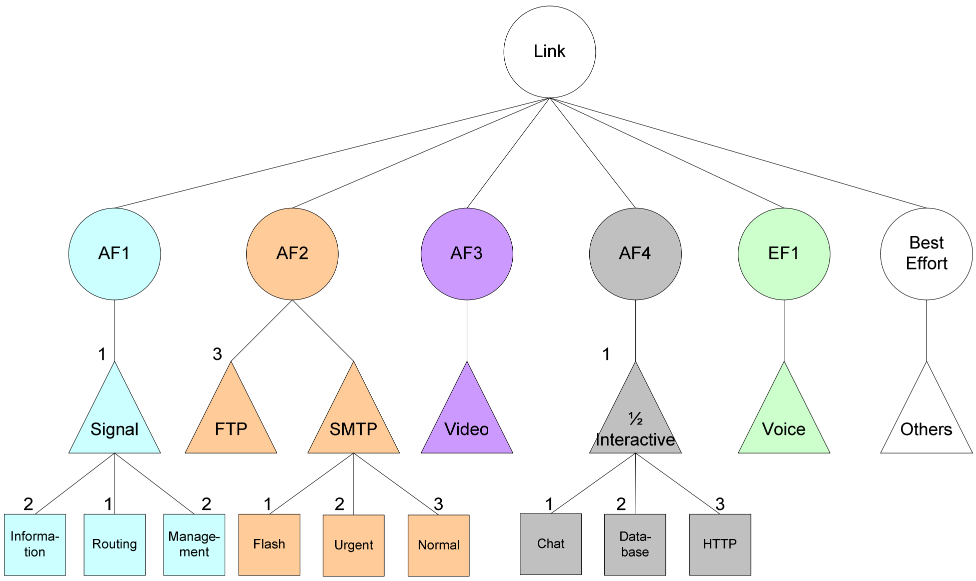


Figure 2.3 DiffServ classification for data traffic in an ad-hoc network. The numbers next to the links represent drop precedence where 1 is low drop precedence and 3 is high drop precedence.

This design might serve as a starting point for our QoS architecture. For the narrow bandwidth waveform we might want to reduce the number of Assured Forwarding (AF) classes. Several studies have shown that a narrow bandwidth network is able to support only a few distinct QoS classes (e.g., [19]).

In the INSC architecture, different priority levels can be handled within each class, e.g. flash, urgent and normal for SMTP messaging. A suggested solution for MLPP for voice within the Expedited Forwarding (EF) class is Multi-Level Expedited Forwarding Per Hop Behaviour [20]. A drawback with this solution is that role-priority and pre-emption is isolated to each traffic class. It is not given that a high priority flow in one traffic class is able to pre-empt low priority traffic-flows in a different class. It should be studied closer how role-priority and MLPP can be best integrated with a small set of DiffServ classes for a narrow band network.

In tactical military network there is likely to be several traffic types where the data has a short lifetime. For these traffic types it could be useful if the network could be informed of the lifetime of a message such that the network can drop stale packets and avoid congesting a link with data that is not needed anymore. A queue type that drops the oldest packet (head drop) instead of the classical tail drop mechanism can be one way to implement such a mechanism. A timestamp with lifetime of the packet is another mechanism, but this timestamp will not be visible for the transport network if IPSec encryption in tunnel mode is used.

A mobile wireless network with a common channel is very sensitive to network load. High network load leads to higher probability of packet collisions (dependant on the chosen MAC algorithm). Admission control is a mechanism that attempts to control the amount of traffic in a network, and thus improves the performance and predictability of a mobile wireless network. Admission control oversees that the total traffic in the network is kept below a certain threshold. This mechanism must also oversee that the admitted flows conforms to the agreed amount of traffic. This might be formalized through a Service Level Agreement (SLA). This method should also have the means to ask sources for admitted traffic to reduce their traffic flow (e.g., change the SLA) to adjust for capacity changes in the wireless network due to topology changes or fluctuations in the channel conditions. Explicit Congestion Notification (ECN) [21] is one of several possible feedback mechanisms. For military networks it will also be required that an admitted flow can be pre-empted to allow higher priority traffic to be admitted to the network. The admission control algorithm must be distributed, and it should adapt to the traffic load.

An item that might be interesting to pursue for the QoS architecture is an element of QoS routing, where the routing protocol carries extra information dedicated for QoS operation and provide different routes (if available) to a destination based on the QoS classification of the data-flow. Multi-topology routing [22] is one example of such a mechanism.

2.3.2 Addressing and routing – unicast and multicast

All terminals should have unique network addresses, enabling loop-free unicast, and multicast, routes. The address allocation should be performed with minimal, or no user interaction. It is foreseen that user roles in the military organisation may have an impact on the addressing scheme, however, role based addressing may be handled by higher layers as well. Hardware based network addresses (one example is the IEEE MAC numbers) is one possibility. Another possibility is to utilise address auto configuration, although this is expected to increase signalling overhead in the networks. The terminals must be configured with a unique network layer address before a valid route can be established and communication can take place. Ad-hoc networking requires that unique address assignment should be automatically done [23].

The routing algorithm(s) should support both unicast and multicast within each ad-hoc network. Terminals/users may be members of more than one multicast group at the time. During radio silence, terminals should keep their multicast group membership(s). In addition a number of more detailed requirements have to be decided:

- Maximum response time
- Reliable multicast
- QoS routing
- Authentication and confidentiality with respect to router signalling

Necessary routing traffic overhead should be minimised. Node mobility should be supported for typical vehicle speeds, including helicopters. The routing should take into account information regarding the quality and capacity of the radio channel and support decentralised network management. The routing should take into account the required service and QoS classes. Information regarding topology/routes should be made available at any terminal interface enabling sensible intra network routing for incoming external traffic. Routers can then make efficient routing decisions based on this knowledge.

The network layer expects to receive some information from the link layer to reduce the signalling overhead and improve the offered service quality to higher layers. The following information should be available:

- Current neighbour list
- Link quality to neighbours
- Current buffer load for the supported QoS classes
- Information about radio silence (status information from network to link)

The network must support multihoming (i.e., efficiently handle simultaneous connection to several command posts with a gateway to other networks or the tactical backbone network).

The protocols Ad hoc on demand distance vector (AODV), Optimized link state routing (OLSR), and Topology dissemination based on reverse path forwarding (TBRPF) have achieved the IETF RFC (request for comments) status. However, the IETF MANET working group are continuing to develop a second generation of OLSR as well as Dynamic MANET on-demand (DYMO), the successor of AODV. Four routing protocols for MANETs are currently under development [24]. The reactive routing protocols investigated are AODV and DSR (dynamic source routing). The two proactive routing protocols investigated are OLSR and TBRPF. Hybrid routing combining proactive and on demand schemes for both stationary and mobile users [25] might be an interesting option, however the actual selection of methods requires a thorough study.

2.4 Link layer

The link layer is responsible for activating, maintaining and deactivating network access. This includes medium access assignments to fulfil a range of QoS requirements from higher layers. The solution should cover a wide range of network connectivity, from a shared broadcast channel to the multihop store-and-forward MANET topologies.

Messages from higher layers should be segmented, if required, to adapt to the various frame lengths and data carrying capacity of the physical layer. This can vary on a packet to packet basis as the physical mode may change both with time and between the nodes depending on the current radio channel and interference conditions.

The link layer often provides error detection and in some cases also error control in form of ARQ is required. The latter feature needs further discussion to obtain efficient system error correction including both the physical layer (hybrid ARQ, FEC), the link layer and the transport layer (TCP).

2.5 Physical layer

A draft standard of the physical layer is available in [26]. The draft describes a continuous-phase coded modulation (CPM) developed by CRC and a quadrature amplitude modulation (QAM) approach developed by Telefunken Racoms. The proposed physical layer is tailored for single frequency 25 kHz operation, however it is foreseen that future evolvement of the physical layer may accompany multiples of this bases bandwidth.

The physical layer protocol data unit (PDU) consists of a preamble used for acquisition, a parameter field for signalling a data field and optional mid ambles. During the *acquisition bursts*, the waveform is constructed according to the picture shown in Figure 2.4.

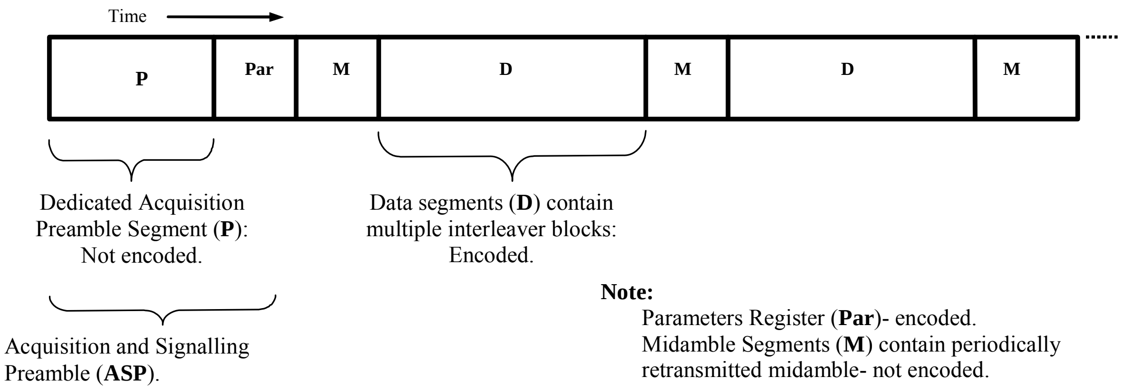


Figure 2.4 Depiction of preamble and midamble construction for fixed-frequency waveform (adopted from [26])

The preamble sequence is followed by the parameter field (PAR), optionally followed by the midamble. The preamble segment (P) and the PAR field are proposed transmitted at a fixed Baud rate of 24.39 kBauds/s according to mode *C2* in Table 2.3. The length of the preamble field is an integer number of a 4 bit uncoded basic sequence 1100. Detection of received preambles is signalled to the link layer. Data segments may optionally be interspersed with a periodically retransmitted midamble. The different physical layer modes are listed in Table 2.3. The parameters *L*, *M* and *H* specify the modulation formats.

Mode	User Data-Rate (kbps)	L	M	H	FEC Code Rate	Baud Rate (kbps)	Spectal Eff.
C1	9.6	1	2	½	½	19230.77	.384
C2	16	2	2	½	2/3	24390.24	.64
C3	19.2	2	2	1/3	2/3	29411.77	.768
C4	28.8	3	2	1/3	2/3	43478.26	1.152
C5	30	3	2	1/5	2/3	45454.55	1.2
C6	32	3	2	1/6	4/5	40000.00	1.28
C7	38.4	3	2	1/8	4/5	50000.00	1.536
C8	48	4	2	1/8	4/5	62500.00	1.92
C9	56	3	2	1/16	4/5	71428.57	2.24
C10	64	4	2	1/16	4/5	83333.33	2.56

Table 2.3 Preliminary waveform modes for fixed-frequency CPM.

The parameter field is scalable and consists of one or more fields containing 24 bits, see Figure 2.5.

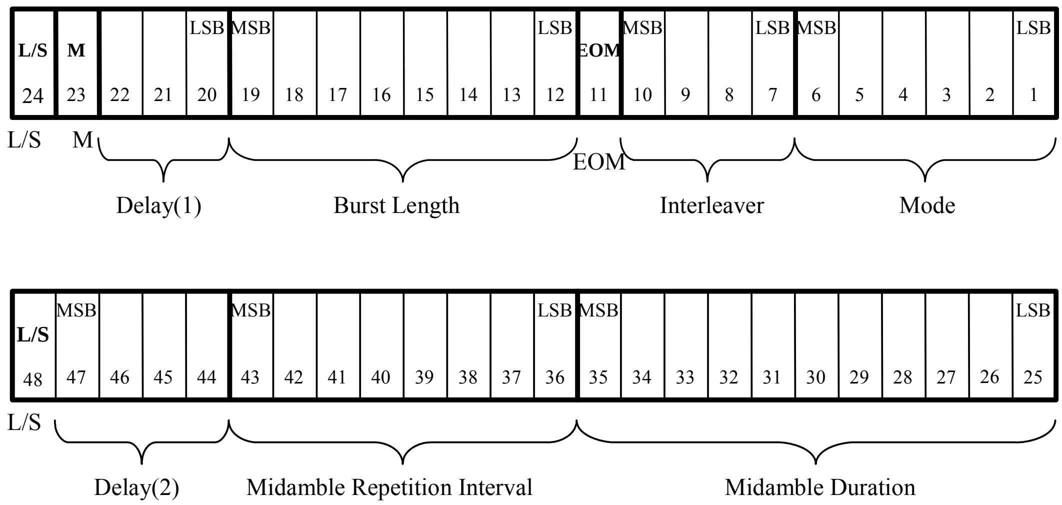


Figure 2.5 PAR register specification (adopted from [26])

The parameter field is protected by forward error correction coding (FEC rate 2/3) and contains information regarding the physical layer mode, burst length, information regarding insertion of midambles and so on. The L/S flag indicates whether the parameter field is extended or not. The minimum interleaver size employed for the PAR field is currently 10 ms at 16 kbit/s information rate, however this might be reduced somewhat if necessary. The performance of the iterative FEC depends on the number of symbols in the interleaver, hence both the interleaver time duration and the symbol rate affects the required SNR to achieve the required bit error rate (BER). The type of service will influence the required BER, or packet error rate (PER), hence a detailed study on required quality (BER/PER) and interleaver length has to be taken into account when designing the MAC layer.

This duration of the PAR field corresponds to 243 coded symbols. With a rate 2/3 FEC, 162 information bits may be transferred, including the 16 bits for CRC. This implies that an acquisition and signalling preamble (ASP) has about 146 bits that may be employed by PAR fields and by for example the MAC layer for signalling purposes. If allocating all of the available information bits to the MAC in the case of just one PAR field, there are 122 bits (15 Bytes) available for every short signalling burst. With a reduced interleaver size of 5 ms, the corresponding number is 41 bits, or about 5 Bytes.

When operating in burst TDMA modus, transmissions are preceded by the basic 24 bit Par register, as shown in Figure 2.6. The duration of the burst is specified in this header.

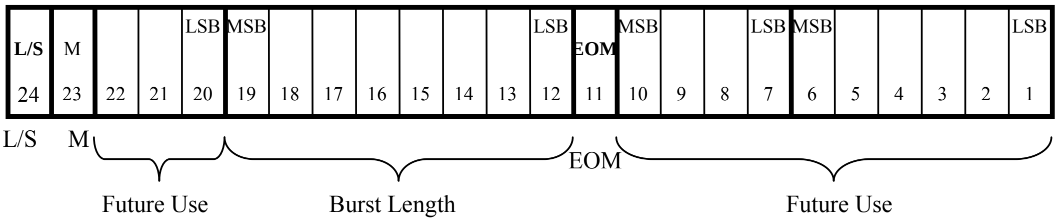


Figure 2.6 Par register during regular TDMA operation.

In this case the modulation of the PAR register is proposed to be identical to that of the current operating mode.

A future evolvement of the physical layer may include scalable wideband modulation and coding for higher capacity networks as well as cognitive radio concepts.

2.6 Radio propagation channel

Norway, and other areas where Norwegian forces are deployed, does have a mixture of flat terrain, mountainous terrain as well as fjords and valleys. There is currently limited detailed knowledge regarding the characteristics of the radio propagation channel and no generally accepted models exist. During the design phase of the Norwegian multi-role radio (MRR) a limited set of VHF measurements were carried out, and some of these do contain some information regarding multipath propagation and delay spread. Significant signal components with a delay in the range of 40 to 60 μs were observed in the area of Kjeller at about 75 MHz. This is a flat area surrounded by ridges [27]. The delayed components had a power 3-4 dB below the main peak, and were sensitive to position and frequency. Similar results were reported in [28], where long impulse responses were measured especially for frequencies exceeding about 50 MHz. In mountainous terrain equal-powered components with a time separation of up to 100 μs were observed. The measurement campaign focused on obtaining examples of channel values suitable for radio design criteria. Although it is not possible to deduce the probability of having significant delays in the order of 50 to 100 μs, it would be of general interest to investigate the performance of the physical layer with such large delay spreads. The investigations of the proposed frequency hopping modulation and coding scheme's sensitivity to inter symbol interference (ISI) presented in [2] consider significantly shorter delays based on measurements in the Ottawa area.

2.6.1 Radio propagation and topology dynamics

The radio propagation channel at VHF (30-300 MHz) and UHF (300 MHz – 3 GHz) frequencies incorporates a range of different propagation mechanisms such as direct propagation, diffraction, reflections, refractions, ground wave propagation, tree attenuation and so on. The strength of the signal will vary both with time and location. For channel modelling purposes 3 components are often employed: path loss, shadowing and fading. The ITU has allocated frequency spectrum to NATO including parts of the range 30 - 400 MHz, which is of main interest for the land mobile services discussed in the current document. For land mobile communications, the band 30 – 88 MHz has traditionally been used, while the 225 – 400 MHz band has been utilised for air-ground-air, satellite and maritime communications. It is expected that this separation of usage types with respect to frequency range will continue to dominate although increasing spectrum congestion during large operations may lead to other frequency assignments. In the future a flexible and dynamic spectrum management approach may be implemented in a cognitive approach, blurring the separation between communication type and operational frequency.

The impact of the radio channel time dynamics on higher layers is discussed in for example [29]. The frequency range 30 - 400 MHz corresponds to wavelengths between 10 – 0.75 meters. Multipath propagation leads to fading on the scale of a few wavelengths on narrow-band channels, and leads to time dispersion of pulses on wideband signals [30]. As a first approximation we assume that the fast fading caused by multipath propagation will be handled by the physical layer forward error correcting codes for vehicular mounted nodes on the move. In this case, fading will normally not cause loss of connectivity and thereby create topology changes. However, for man pack terminals or slowly moving vehicles the fading pattern will be slow relative to the time slot (burst length), potentially leading to rapidly changing network topology. This is largely dependent on whether a direct component between the transmitting and receiving node exists, or not. In the former case the envelope has a Nakagami-Rice distribution, and in the latter case the Rayleigh distribution describes the envelope variations, given that a large number of propagation paths exist. Jakes classical Doppler spectrum might be utilised to describe the spectral form of the received signal, and thereby the time/distance fast dynamics of the channel [31].

Urban measurements reported in [32] indicates that the surface wave may be neglected for transmitter antenna heights above 0.85 m in the frequency range 225 – 400 MHz, and that a two-ray model taking into account a direct and a reflected component. Values for the path loss exponent near the transmitter (n=2), as well as at longer distances (n=4), are given together with the corner loss utilised to describe the transition between the two propagation regions. In the current study it is assumed that the path loss changes relatively slowly compared to the time slot durations, and that topology updates due to path loss variations occur relatively seldom. An exception is probably ground-air communications where distances vary quickly.

Most topology changes for vehicular mounted nodes on the move are expected to be induced by time varying shadowing from obstacles such as buildings and local terrain features. Field-strength values in the shadows of many small obstacles follow a lognormal distribution [30] when the fast fading is filtered out. The median of the distribution can be estimated by deterministic methods, while the standard deviation, in the typical range between 3 and 10 dB, can be estimated utilising methods described in [33]. The large scale shadowing variance has been extracted from measurements in for example [34], reporting 8 and 12 dB in "regular" terrain for VHF and UHF respectively.

Although the above discussion is somewhat limited and do not quantify the topology rate of change, it seems clear that shadowing and multipath fading may cause rapid changes in connectivity between the nodes. This is of major importance when considering TDMA scheduling algorithms, routing methods as well as retransmission methods. Retransmissions at the link layer will decrease the frequency of topology changes and is foreseen to be utilised especially for non real time services. The requirement for changing the routing paths is thereby a combination of acceptable traffic delivery delay and system overhead, and a time based hysteresis should be applied to ensure that only transmissions problems at the two lower layers lasting longer than a limit results in route changes. A challenge is to adjust such a hysteresis to the combination of traffic types and various link unavailability statistics.

2.6.1.1 Multipath fading

Fast fading due to multipath propagation is probably the fastest varying factor that may have implications on network topology. It should be noted that rerouting based on fast fading can result in too much routing overhead, and that link layer retransmissions with an acknowledge scheme can be utilised to cope with these dynamic channel events.

The diffuse signal component causing fast fading often has a Rayleigh distributed envelope, given that there exists a significant number of locally reflected signal components of comparable strength around the receiver. At VHF a Nakagami-Rice distributed envelope may well describe the propagation more realistic due to the existence of a direct (surface) component, thus Rayleigh fading represents a worst case. The probability density function (PDF) of the instantaneous signal power ($r^2/2$) when the mean power is a^2, is for the Rayleigh distribution given by

$$P_r(r) = \frac{r}{a^2}\exp\left(-\frac{r^2}{2a^2}\right) \tag{2.1}$$

Let the normalised envelope be denoted as ρ. It is given by

$$\rho = \frac{R}{R_{RMS}} = \frac{R}{a\sqrt{2}} \tag{2.2}$$

Hence, the CDF for the normalised envelope is given by

$$P(\rho \le P) = 1 - e^{-P^2} \quad (2.3)$$

The normalised complementary CDF of the envelope is shown in Figure 2.7.

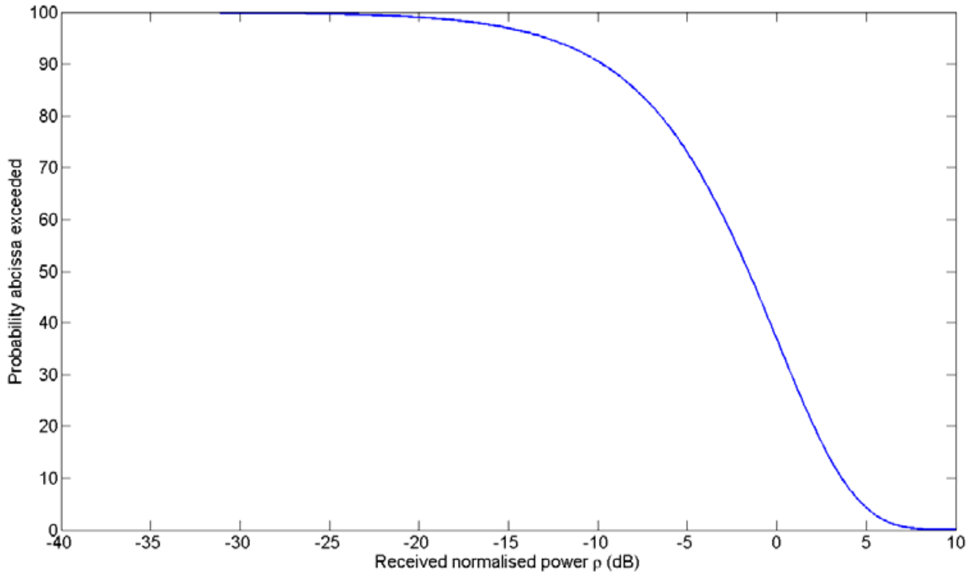


Figure 2.7 Complementary Rayleigh CDF as function of normalised envelope.

For a fading signal, the average fade duration (AFD), *T*, is by definition the average time over which the signal envelope, *r(t)*, remains below a certain level [35]:

$$T_{Fade}(R) = \sqrt{\frac{a^2}{\pi}}\,\frac{e^{\left(\frac{R^2}{2a^2}\right)} - 1}{Rf_d}, \quad T_{Fade}(\rho) = \frac{e^{\rho^2} - 1}{\rho f_d \sqrt{2\pi}} \quad (2.4)$$

were f_d is the maximum Doppler frequency spread. This is the result of dividing the cumulative density function by the level crossing rate (LCR) function; a resulting plot is shown in Figure 2.8.

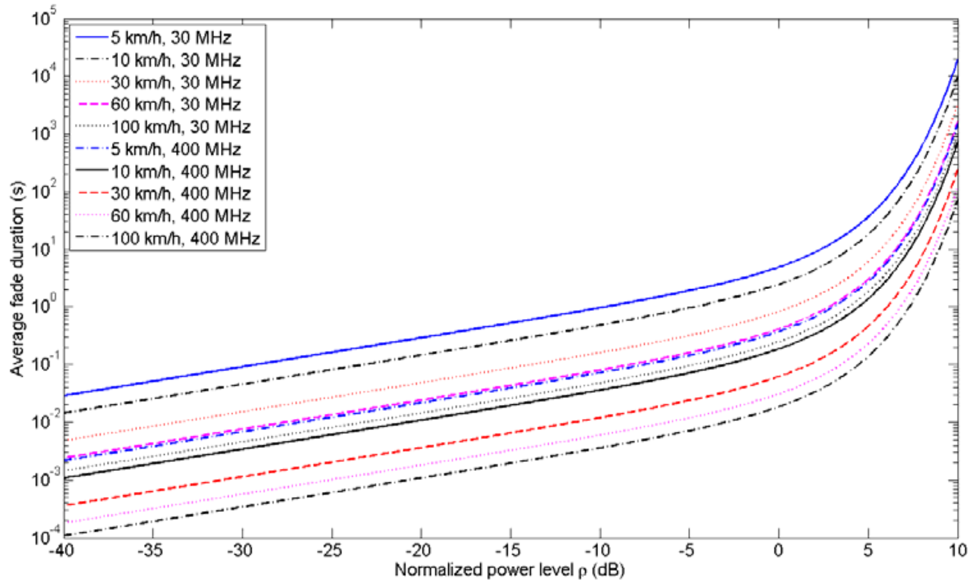


Figure 2.8 Average fade duration for the Rayleigh distribution, normalised envelope

To obtain the inter-fade distribution, it is viable to divide the complementary CDF function with the LCR function resulting in

$$T_{InterFade}(\rho) = \frac{1}{\sqrt{2\pi}\, f_d} \tag{2.5}$$

The resulting inter-fade distribution is shown in Figure 2.9.

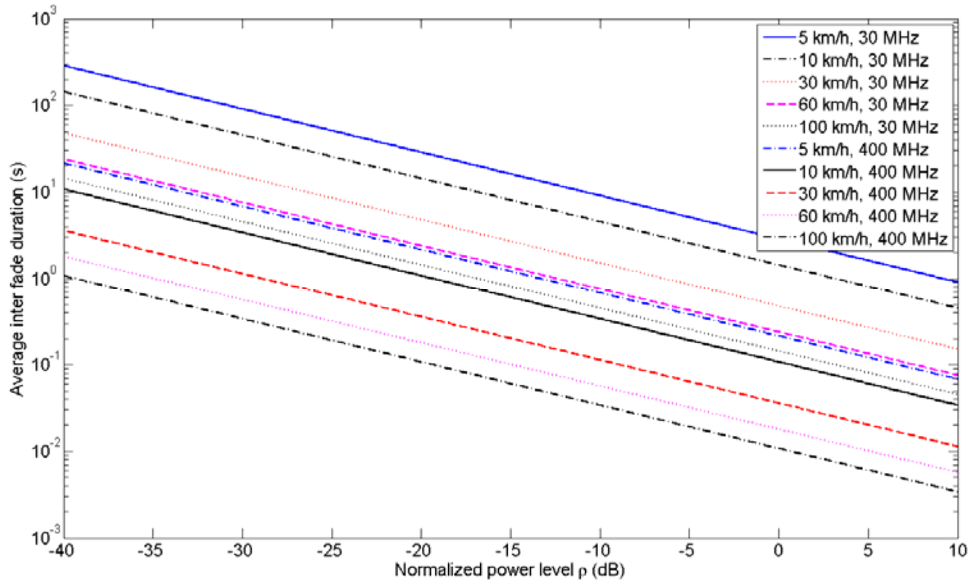


Figure 2.9 Average inter fade duration for the Rayleigh distribution, normalised power

It may be discussed whether a Rayleigh distributed envelope is a representative model for the MANET network. At the lowest frequency it is probably most often not the case, as large reflecting surfaces (relative to the wavelength) surrounding the receiver requires large obstacles such as mountains and buildings. However, this type of operational environment is quite typical in part of for example Norway, Iraq and Afghanistan.

If we assume a normalised power margin of 20 dB, the average fade duration for the Rayleigh distribution is in the range of 1 - 300 ms, with a typical value of perhaps 15 ms for VHF vehicles travelling relatively fast and UHF man-packs at walking speeds. The average inter fade duration ranges from 0.1 - 30 s, with a corresponding typical value of about 2 s. This duration of time between fades is relatively long, and if several frames is accommodated between fades the rate of topology change seem to be within an acceptable limit with respect to overhead generation.

3 Link layer

The access to the radio channel shared within the ad-hoc network is managed by the MAC protocol which is part of the link layer. A good overview of early packet radio MAC algorithms for fully connected networks is given in [36]. A more recent overview of MAC algorithms is given in [37], including QoS support for different types of services. Fixed and random access techniques as well as a few adaptive solutions combining distributed TDMA and random access are discussed. The main goal of the adaptive solutions is to obtain the low delay experienced at low loads for the random access schemes combined with the stability and low delays at high network loading for the on-demand resource allocation algorithms. Furthermore, the performance degradation of Carrier Sense Multiple Access (CSMA) when experiencing hidden terminals is highlighted. The flexibility required in our design solution does require a solution working properly even when there is a reduced connectivity degree requiring multiple hops, although in many cases a near full degree is expected to be obtainable at VHF frequencies

MAC protocols can be divided into three main classes according to [3]:

- contention based protocols
- allocation based protocols
- hybrid protocols combining contention and allocation

A thorough discussion of TDMA allocation protocols may be found in for example [38].

3.1 Contention based MAC protocols

Contention protocols utilise direct competition to determine access rights. Examples of contention MAC protocols include carrier sense multiple access (CSMA) in various flavours. CSMA suffers from hidden node interference whenever there is reduced connectivity within the network and thereby becomes unstable at high loads. The hidden terminal condition requires that transmission collisions occur undetected by the transmitting terminal. Handshaking routines utilising for example request to send (RTS) and clear to send (CTS) address the hidden node interference, however, the challenge of instability at high network load remains. An alternative to CSMA collision detect (CSMA/CD), that attempts to reduce the penalty from hidden terminals, is multiple access with collision avoidance (MACA) [39]. MACA is not utilising the carrier sense methodology and may be extended to include power control. Both CSMA and MACA protocols are asynchronous and do not require an accurate timing reference for the terminals. Random access protocols such as Aloha, slotted-Aloha and spread-Aloha may be included in this class of MAC protocols as well.

The currently considered MANET requires QoS guarantees for the voice service, and possible other time sensitive services as well. Contention protocols do not normally provide this possibility, nor stability under heavy load conditions.

3.2 Conflict-free MAC allocation protocols

Allocation protocols, such as TDMA, utilise synchronised time slots where only one terminal is allowed to transmit at any given time. Hence, a scheduled collision free access is obtained at the expense of reduced efficiency for bursty random traffic and perhaps slower adaptivity to changing traffic conditions. Allocation is one method to guarantee QoS and therefore an interesting concept when transporting real-time traffic such as voice.

Early investigations of conflict free allocation include alternating priorities (AP), round robin (RR) and random order (RO) [40]. These latter methods are effective only for a moderate number of users and do not scale very well. The mini-slotted alternating priorities (MSAP) is one alternative to reduce the overhead as the number of users increases.

Most of the available allocation algorithms require global connectivity information and are thereby of the centralised type [41]. In an environment with fast changing topology, decentralised or distributed algorithms are required to reduce the required management overhead and cope with the changing topology in time. The distributed broadcast algorithm proposed in [42] - [43] requires two-hop connectivity information and 4 hop scheduling information, but adaptivity to changing topology remains to be developed. The broadcast algorithm described in [44] requires information regarding the nearest neighbours only. According to [41], the efficiency and robustness of these algorithms are somewhat questionable in MANETs.

3.2.1 Spatial/Dynamic Time Division Multiple Access

Reservations in dynamic TDMA may be either centrally coordinated by an elected leader node or distributed. Early works on wireless ATM schemes [45], an extension of the PRMA protocol [46] and IEEE 802.15.4 [47] are all examples of the centrally coordinated variants. In this case nodes contend in the start of a super frame and the leader node (base station) confirms successful reservations with assignment of slots. Such a scheme seems effective for traffic flows with sufficient duration such as voice and larger data messages, while shorter data packets may just as well apply for example slotted Aloha without explicitly setting up a reservation. Distributed variants typically include a request-grant-confirm scheme as in for example [48]. The slot allocation methodology may either be based on a distributed scheme, or by selecting cluster leaders functioning more or less as centrally controlling base stations regarding the signalling. Examples of the latter are discussed in for example [49].

Spatial Time Division Multiple Access (STDMA) was introduced in [50] to avoid collisions in a multihop network. The distribution of time slots requires knowledge of the connectivity in the network and identification of so called cliques, which closely resembles the topology information required in some routing protocols. The main idea is to reuse time slots whenever the interference conditions are not to severe, requiring either global or local information regarding the signal to interference plus noise (SINR) conditions. Due to the possible rapidly varying radio channel and traffic patterns, both the signal level from the transmitting node, and the interference levels from the distant nodes sharing the same time slot might impose frequent updates in the scheduling scheme. A discussion related to the required overhead for distributed STDMA with an interference based scheduling scheme is given in [51]. The report is the currently last in a series of STDMA related work on MANETs and discusses the type of information required for interference based scheduling in a distributed STDMA mobile network. The required overhead may be transmitted in mini slots prior to the traffic data with each node assigned one mini TDMA slot, see Figure 3.1. Thus the number of mini TDMA slots grows linearly with the number of nodes in the network. With a time duration of for example 6.2 ms per mini-slot at 16 kbit/s, a 25 node network would require 155 ms per frame for signalling. The corresponding number for a network with 100 nodes is 620 ms. It is clear that to obtain a reasonable efficiency the mini slots need to be transmitted relatively seldom. This is in contrast to the resource reservation algorithm requiring the information in the mini slots to reserve time slots for active nodes. It may be possible to do this process adaptive, however, a significant time delay is expected when silent nodes enter an active state with information to be sent.

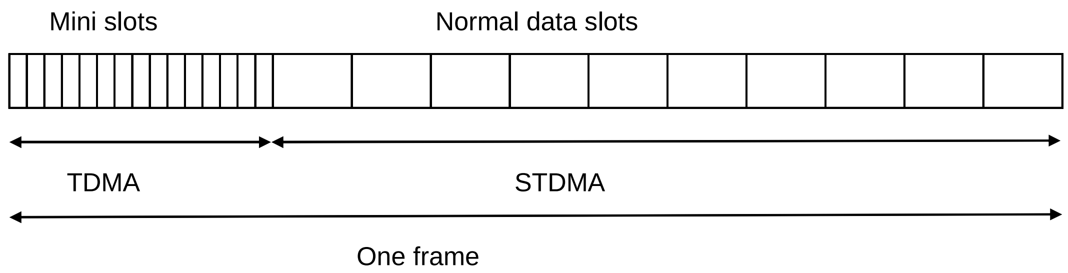


Figure 3.1 Spatial TDMA frame format

A discussion on assignment strategies and the impact on throughput at various load conditions is given in [52]. STDMA is most efficient if the network does not change too quickly, since changes may force an update of the schedule. Traffic sensitive scheduling algorithms allocating more slots to nodes with high priority (due to for example long transmit buffer queue) outperform non traffic sensitive schemes [53], [54]. Combinations with rate adaptation and power control are discussed in e.g. [55]. The assignment of time slots may be either based on nodes or links, where the former allow a node to transmit to any recipient node while the latter assign time slots to specific transmitter-destination node pairs. The efficiency with respect to throughput of the two approaches depends on the selected frame length, and a combined assignment scheme is proposed in [56].

In the following two maritime ad-hoc networks employing STDMA variants are briefly discussed and a few points made between the low mobility maritime environment and the potentially high mobility land-based MANET for mixed services currently under study.

3.2.1.1 Case study: Sub net relay

The Subnet Relay (SNR) is a MANET targeting maritime operations. FFI has previously performed initial investigations of the performance of this network [57]. The MAC algorithm is (by Rockwell Collins) called distributed slot reservation media access (DSRMA). The DTDMA scheme shares channel bandwidth on a single frequency amongst the network nodes. Timing information is taken from GPS or another stable time source. The following description is closely related to the information given in [58]. Time on the shared channel is divided into cycles which are further divided into time slots. Each participating node reserves one or more time slots in a cycle for transmission to one or more of the nodes. Each cycle also includes one or more random access slots which are not usually reserved but in which nodes without a reserved slot (such as new joiners) can request a regular slot for transmission. Under normal circumstances, each node will maintain at least one slot per cycle for a broadcast transmission to all of its neighbours. With DSRMA, nodes inform their neighbours of slot ownership within their two hop neighbourhood. This allows nodes to avoid collisions by not transmitting at the same time within the two-hop neighbourhood. Slot re-use occurs naturally under this scheme as 3 hop neighbours are unaware of each other’s slot usage and so those slots can be reused.

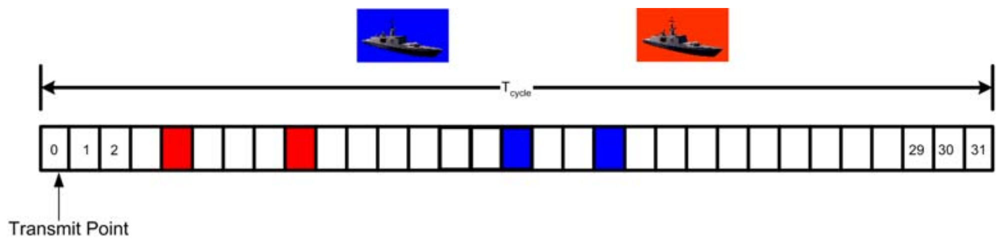


Figure 3.2 Illustration of slot allocation in Sub net relay

Slots are assigned according to traffic load and other factors and can be pre-allocated and merged to increase efficiency. In general, as buffers fill, a node will request more slots and as buffers empty, a node will release slots. Member nodes obtain ownership of slots for transmission of data to their one-hop neighbours.

Sub net relay allocates at least one traffic slot in each frame/cycle per node. Thus, the design is optimised for a network with few nodes and it is not very scalable. A mini-slotting TDMA variant could thus improve the network scalability, enabling a higher number of nodes in the network.

3.2.1.2 Case study: Automatic Identification System

STDMA is utilised in the maritime anti collision and identification system AIS. AIS is an international standard for ship-to-ship, ship-to-shore and shore-to-ship data communication, including vessel position, speed, course, destination and other data defined by ITU-R Rec. M.1371-1. AIS operate at VHF frequencies and utilise accurate timing obtained from for example a satellite navigation system. In AIS, the STDMA frame is one minute long. The frame is divided into time slots carrying user data. A variant of STDMA, denoted as Self Organizing Time Division Multiple Access (SOTDMA), is employed allowing a large number of nodes to share a narrowband channel by synchronizing their data transmission to an exact timing standard [59].

3.2.2 Orthogonal frequency division multiplexing

Orthogonal frequency division multiplexing (OFDM) is a popular modulation method utilised in mobile networks with significant channel delay spread. OFDM is interesting seen from an anti-jamming perspective where frequency hopping by utilising different sub-carriers can be utilised. By scaling the number of sub-carriers, the bandwidth can be matched to the available spectrum and the capacity requirements for both wide- and narrowband approaches. Normally the peak-to-average power ratio is relatively high in OFDM, however, constant envelope approaches are reported in the literature enabling efficient utilisation of a terminals power amplifier [60]. MAC approaches suitable for MANETs are currently under study. For example a random access approach is given in [61], while a dual busy tone approach is described in [62]. In the latter approach, the control channels utilised by the hybrid MAC approaches described in Section 3.3 are mapped onto sub-carriers in the frequency domain.

3.2.2.1 Case study: WiMAX

WiMAX is a wireless broadband access system based on the IEEE 802.16 standard. It is expected that some WiMAX equipment will incorporate the mesh mode operation, covering both centralised and distributed MAC algorithms. In the mesh mode nodes utilise a pseudo-random function to compete for their transmission opportunities based on the scheduling information available in their two-hop neighbourhood [63]. Data sub-frames are allocated through a request-grant-confirm handshaking procedure. The data-frames and control sub-frames are separated, and the transmissions in the control channel are scheduled to be collision-free. Multiple data channel slots may be reserved.

3.2.2.2 Case study: HiperLan 2

HiperLan 2 is the successor of HiperLan 1 and is wireless local area network standardised by ETSI [64]. Commercially the IEEE 802.11 standard (WiFi) is significantly more widespread, however, the ETSI approach is of interest due to utilisation of dynamic TDMA combined with orthogonal frequency division multiplexing (OFDM). The offered bit rates is within a range of typically 6 - 54 Mbit/s. HiperLan 2 operates in a centralised mode to provide typically multimedia and data services to end users connected to an access point. In addition an ad-hoc mode intended for residential utilisation is provided enabling direct link communications without the support of a cellular network infrastructure. In this case a central controller (CC) is dynamically selected amongst the user nodes, providing QoS support similarly to the case where access points are involved. In this direct mode a centralised MAC protocol is used, while the user traffic is flowing directly between the nodes.

3.2.3 Topology-independent MAC protocols

Allocation TDMA schemes require updates of the local or global neighbourhood for assigning slot access rights to the nodes. If the network topology changes often, significant signalling overhead in terms of neighbour updates occurs even for the distributed algorithms. On the other side, the bandwidth efficiency of a topology-transparent schedule is lower than that of a topology dependent schedule due to the inherent redundancy in order to work topology-independent [48].

3.2.3.1 Time spread multiple access

Variants of topology-independent scheduling algorithms denoted as *time spread multiple access* (TSMA) are discussed in [65] – [68]. The frame, L, is divided into q sub frames, each with q slots, resulting in q^2 slots per frame, and $L = q^2 << N$ where N is the number of nodes in the network. Each node attempts transmission q times per frame. The algorithms guarantee that each node has at least one collision-free time slot in each frame, thus limiting the maximum time delay and enabling QoS time delay guarantees.

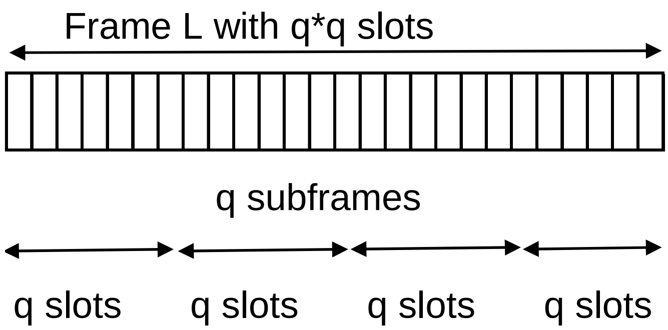


Figure 3.3 Time spread multiple access frame format

The nodes may transmit in a number of additional slots, but these may be perceived as contention based slots. The basis for the algorithms is selection of individual transmission schemes depending on the number of nodes in the network, and the node degree. The degree of a node, D, is defined as the number of nearest neighbours, a quantity that will be time dependent in a mobile ad-hoc network. Two requirements that have to be fulfilled if the TSMA performance is not degraded: $q \geq kD + 1$ and $q^{k+1} \geq N$, where k is a positive integer and N is the number of nodes. Normally a frame length of $L = q^2 << N$ should be selected to obtain performance improvement over static TDMA. Network throughput and delay is to some degree dependent on that the maximum degree is estimated correctly. Hence, protocol threading was developed in [69] to remove the requirements on estimation of the maximum node degree at the expense of longer time delays.

In the original TSMA approach denoted proper robust scheduling (PRS) [65], the nodes transmitted the same information throughout the frame at each of these transmissions. In [70], it is indicated that the average throughput performance of TSMA is at least as good as Slotted Aloha, without the degradation at high loads as seen in Aloha protocols. An acknowledge scheme was developed in [71], allowing the transmitting node to send new information in the successive time slots if positive acknowledgements are received. If the transmitting node has no more information to send, it halts transmission enabling other users to utilise the time slots with a lower probability of collision and thereby increase throughput at high traffic load.

As the TSMA protocol efficiency depends on the number of nearest neighbours, the fairness of the protocol, and the variant with acknowledgement, is not well developed. A p-persistent transmission scheme, including the acknowledgement feature, is described in [68]. The value of the transmission probability depends on the number of available collision free slots, and hosts with few collision-free slots (many neighbours) have a higher probability of accessing the contention slots. The efficiency of TSMA compared to CSMA utilised in a control channel is discussed in [71], where CSMA is found to outperform TSMA.

In TSMA, the delay bound grows logarithmically with network size, N, and quadratically with the maximum node degree D. It is possible to limit the node degree by employing, for example, adaptive power control to limit the number of nearest neighbours when node density is large. In [72], a power control algorithm is described, utilising neighbour information from the routing table to adjust the local number of nearest neighbours and thereby optimising the TSMA performance with respect to throughput and delay. One side effect of the approach is somewhat reduced battery drainage, important for man pack terminals as well as better resistance towards varying interference and noise levels. Another effect is the coupling between routing and connectivity as described in [73]. Lowering the transmit power should not partition the network, nor generating significantly more overhead from routing updates. The selection of power control algorithm is dependent on the chosen routing method, where for example link state protocols provide global routing information to each node.

TSMA will typically perform well in a network with rapid topology changes and limited connectivity degree. This could be for example vehicular troops on the move in rough or build up terrain, where especially the UHF frequency range would suffer from frequent connectivity changes. A situation where TSMA would have to be operated in a degraded region is stationary collection of troops in for example a camp or in a column where it would be difficult to limit the degree of connectivity. In such cases dynamic TDMA, for example, would probably outperform TSMA.

3.3 Hybrid MAC protocols

Hybrid protocols combine the features of contention and allocation MAC protocols. These protocols are designed to avoid instability and still adapt to changing capacity requirements in the network.

3.3.1 Distributed packet reservation multiple access

Packet reservation multiple access (PRMA) normally has a centralised structure, but it can be modified for MANET usage, denoted as distributed PRMA [74]. The modifications proposed in the paper is tailored for voice coders with variable bit rate, including periods with silence. Thus, the reservation is performed on a packet basis without setting up a continuous resource reservation. It should be noted that the voice coder assumed in this study delivers constant bit rate irrespective of whether one speaks or not and very frequent connection setups are not expected for multicasted voice. The approach takes into account hidden and exposed terminals.

The channel is divided into equal length frames, see Figure 3.4. The frames are further subdivided into slots carrying both signalling and user data traffic. The slots are divided into mini-slots utilised for contention purposes when the slot is not reserved. The signalling part of each slot consists of two fields: request to send/busy indicator (RTS)/BI and clear to send/busy indicator (CTS)/BI where BI is a busy indicator. These fields are used both for slot reservation and to prevent collisions due to hidden terminals. A node transmits an RTS in the first available (idle) mini-slot. This implies that the first part of each RTS slot is dedicated to channel sensing. If the receiver receives the request, it responds with a CTS message in the same mini-slot. Then, if the transmitter receives the CTS message, it may use the remaining part of the slot for sending data traffic immediately, thereby stopping further contention attempts. If a CTS is not received successfully, the contention process continues in the next mini-slot in the same slot. If a successful reservation is made by a node, the same slot may be used in the subsequent frames until the end of the flow of packet transmission.

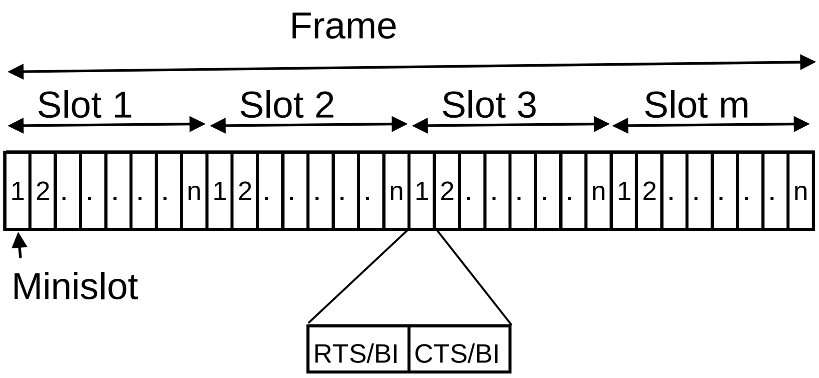


Figure 3.4 Frame structure for D-PRMA

The PRMA approach implement prioritisation between voice and data by allowing nodes with voice traffic to contend in the first mini-slot with a probability equal to one. Data traffic is allowed to transmit RTS in the first mini-slot, but with reduced probability for placing the request in the first mini-slot. Only voice terminals are allowed to reserve the subsequent slot position in the next frames. To improve the performance of data transmissions a similar piggybacked reservation mechanism for the subsequent frames could be implemented for data traffic as well. In addition a call admission control mechanism limiting the number of active nodes may improve the throughput by reducing contention collisions. The latter mechanism is foreseen to be important if variable rate voice coders are utilised and the reservation process is performed for each voice burst since the reservation is lost during the silence periods. Utilisation of mini-slots for short data messages such as position information may also improve the performance for the MANET under study.

Pre-emption will require an additional mini-slot, used either for prioritised contention access or for signal dropping of reservations for slots already reserved. This can probably be implemented similarly to the prioritisation mechanism by utilising the first mini-slot for high priority traffic.

The solution to hidden and exposed terminals should prevent contention in following mini-slots if one node wins the contention in a previous mini-slot. These cases are handled by utilising the RTS/CTS dialogue as well as the busy indicator BI. When a voice transmission terminates, the receiving node stops sending in the BI field. In addition, contention in slots already reserved in the subsequent frames should be avoided.

3.3.1.1 Contention for traffic flow reservation

We want to derive the probability for a successful transmission of RTS/CTS in an available slot, *T(n)*, given *n* contending nodes available at the beginning of the slot and follow the calculations in [74] relevant for unicast in distributed PRMA. If $n > 1$, there will be a collision in the first mini slot and the *n* terminals contend in the *m* extra mini slots since all nodes by default contend in the first mini slot in the PRMA scheme. Thus *T(n)* is equal to one minus the probability that none of the *n* contending nodes has a successful contention in the m extra contention attempts.

$$T(n) = \begin{cases} 0, & n = 0 \\ 1, & n = 1 \\ 1 - [1 - Q(n)]^m, & n > 1 \end{cases} \tag{3.1}$$

where *Q(n)* denotes the probability of a successful transmission of an RTS among the *n* competing nodes. Assume that the nodes select an additional contention slot uniformly between 1 and *m* with probability *p = 1/m*. This Bernoulli trial may be described by the Binomial distribution:

$$Q(n) = \binom{n}{1} p(1-p)^{n-1} \tag{3.2}$$

The probability of successful reception of an RTS is displayed in Figure 3.5.

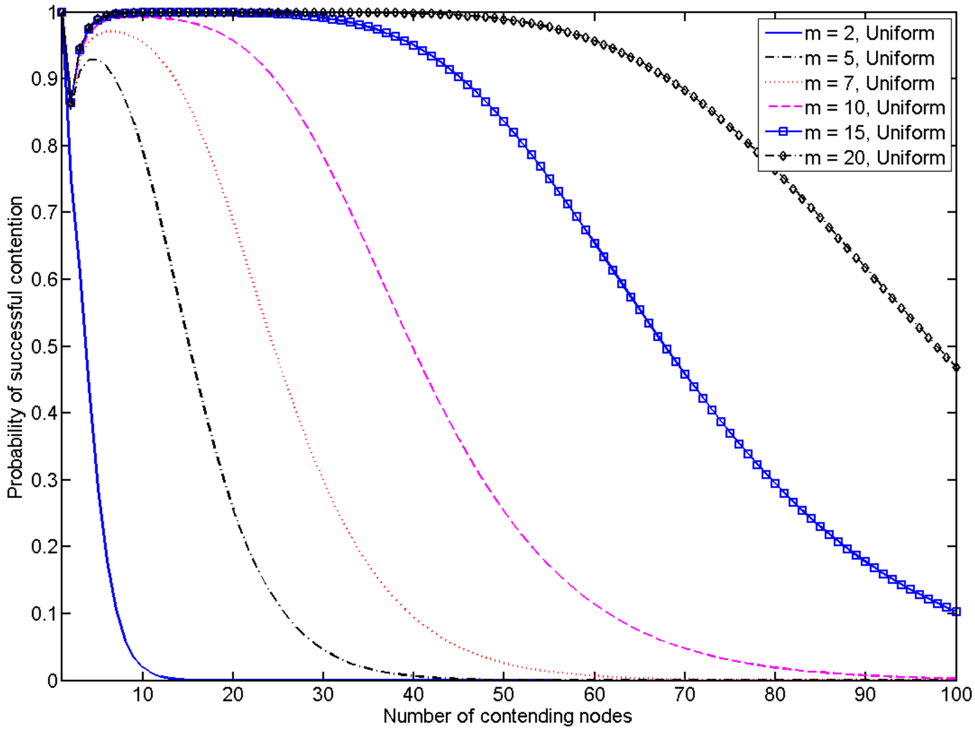
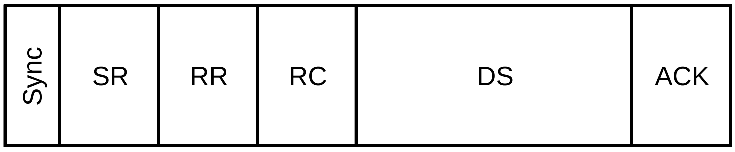


Figure 3.5 Probability of successful contention in distributed PRMA

As seen in Figure 3.5, about 10 extra mini slots available for contention if the first contention fails would give a reasonable chance of success for a CNR network of typical size with high traffic load. If fewer contention slots are available, other means need to be utilised to reduce the probability of simultaneous contention attempts. For very large networks alternatives to the uniformly distributed waiting time before transmission might be useful as the number of extra mini slots required for a probability of successful contention increases to about 40 for 256 competing nodes as the worst case. Although the average number of active terminals who have packets in their transmit queue may be limited during normal operation, events may trigger many terminals to try to transmit information leading to instability of the reservation contention. Connection admission control is thus a prerequisite for the soft reservation mechanisms, probably combined with prioritised contention in different slots where high priority messages contend alone.

3.3.2 Soft reservation multiple access with priority assignment

The soft reservation multiple access with priority assignment (SRMA/PA) protocol supports both real and non-real time services with QoS support. Soft reservations are performed with RTS/CTS like contention handshake routine. The protocol enables pre-emption and also includes an acknowledgement field in the mini-slot [75]. The frame contains a fixed number of slots, and the slot structure is depicted in Figure 3.6.



SYNC: Synchronisation	SR: Soft reservation
RR: Reservation request	RC: Reservation confirmation
DS: Data sending	ACK: Acknowledgement

Figure 3.6 SRMA/PA slot structure

The soft reservation method implies that nodes with real-time services may “steal” slots form nodes with non real-time services, hence voice is prioritised over data traffic. The soft reservation (SR) field is utilised to avoid collisions and plays the role as a busy-tone indicator. The field also contain information regarding the priority. If no transmission in the SR field occur, nodes may transmit reservation request in the RR field. The RR field is also utilised for pre-emption of for example voice traffic overtaking data slots according to a defined service class priority scheme. When a RR field is successfully received the intended receiver transmits a reservation confirmation in the RC field. The handshaking process thus mitigates both the hidden and the exposed terminal problem.

The initial priority value assigned to a node is determined based on service class, that is, for example voice and data. The priority of reserved nodes will be assigned a priority where $pri_{voice} > pri_{data}$. If reservation in the contention slot(s) fails, the priority is updated according to the urgency of the information to be transmitted. If a user initiates a voice call, the terminal start contention with a priority larger than any ongoing data transfers (as indicated in the SR field of the ongoing data transfer). A proposed algorithm for updates of the priorities in case of reservation collisions are given in [75]. In addition to increasing the priorities, an exponential back off algorithm is implemented to resolve reservation conflicts between terminals with the same priority, for example two voice calls.

The ACK field is mainly intended for TCP communications where fast link layer acknowledgements may be better suited to the fast varying channel conditions than TCP acknowledgements above the network layer. For UDP traffic, or the voice service, the ACK field is not strictly required and it should thus be an optional field. If it is not inserted it could be utilised to for example increase the data burst length. This could be signalled within one of the parameter fields in bursts sent from the node transmitting information.

Information such as geographical position updates could either be piggybacked with user data, routing protocol information transmitted at a regular interval, or be assigned a third priority.

Multicast enabling for example group communications has not been thoroughly covered in the discussion yet. As pointed out in [76] the RTS/CTS type of multihop networks lacks an established and efficient link layer method for multicasting. This is a rather serious problem as group communications are foreseen to dominate the usage of the network. The main challenge is avoiding nodes that are hidden to the originating multicasting node to transmit in the multicast slots and thereby hindering receiving nodes to receive the multicasted message.

3.3.3 Collision avoidance time allocation

The collision avoidance time allocation (CATA) protocol resolves the hidden node problem and is designed for both uni-, multi- and broadcast [72]. The CATA approach divides a slot into five mini-slots as shown in Figure 3.7.

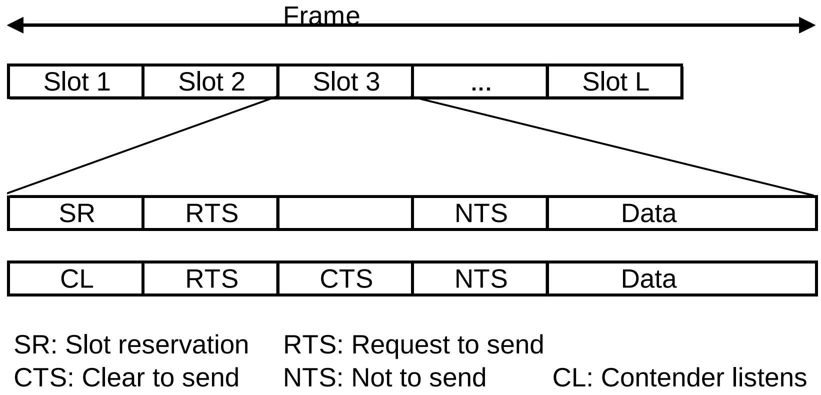


Figure 3.7 Slot and frame structure of CATA (Adopted from [72])

Unicast
A node receiving data sends a slot reservation (SR) packet in the first control slot, serving as a busy tone. In addition, any node sending data in the slot sends a request-to-send (RTS) during the second control slot to jam any RTSs sent to its neighbours from nodes who may not notice the reservation. Both the transmitting and the receiving node keep quiet during the third control slot after soft reservation, and the transmitting node sends a not-to-send (NTS) message in control slot 4. During the reservation phase, a node receiving an RTS intended for it self sends a clear-to-send (CTS) message in control slot 3. If the reservation is successful, data can be transmitted in the current slot and in the same slot in subsequent frames until the unicast transmission is terminated.

Multicast
The originating node send a broadcast RTS in control slot 2. All nodes that correctly receives the RTS, or do not hear anything in the second slot, remains silent in control slot 3 and 4. If any of the nodes detect a signal in the second slot that it is not able to decode, it sends an NTS message in control slot 4 as a negative acknowledgement to any potential broad- or multicast reservations being made. If the node with the group message to send detects an NTS message in control slot 4, or detects noise in this slot, it defers the reservation and has to make a new attempt in another slot or at a later stage. If the fourth control slot is clear, however, reservation is successful and the multi/broadcast flow is started.

Instead of a ACK field for the data as in SRMA/PA, a negative acknowledge control burst is inserted before the data burst to avoid collisions during broad- and multicast. This implies that the signalling overhead is about the same in the two protocols. CATA resolves the multi- and broadcast issue, which is of major importance in the targeted MANET. The pre-emption feature of SRMA/PA is not directly feasible in the original CATA protocol as the transmitting note sends an RTS in every slot to hinder any hidden nodes to the receiver (who is the one transmitting the busy tone SR field) to contend for the channel. The NTS field might be utilised for this purpose as well.

The contention challenge is similar to distributed PRMA, with only one RTS/CTS possibility in available slots. This implies significantly longer waiting times compared to the case of several mini slot contention possibilities. It is expected that such a network employs a more sophisticated collision resolution or back off strategy to improve throughput during heavy reservation load and to implement priority access.

3.3.4 Alternative hybrid protocols

Collision-free Receiver-Oriented MAC (CROMA) is a receiver oriented TDMA based collision-free MAC protocol handling both the hidden and the exposed terminal problem as well as both uni- and multicast [77]. It requires time synchronisation, is collision-free and receiver oriented. ADHOC MAC [78] is an interesting protocol enabling QoS provisioning dynamic TDMA in the form of reliable reservation-Aloha (RR-Aloha). The protocol is designed for environments without energy limitations, for example inter vehicle communications, and requires continuous node transmission. It is unsure how the protocol performs with battery driven terminals where the power consumption should be limited.

3.4 Physical layer adaptivity

Adaptive power control may increase capacity in MANETs [79]. It can be actively used in topology control as discussed in conjunction with topology independent MAC protocols.

The network topology is also affected by the selected modulation and coding scheme. There are important differences between power control and adaptive modulation and coding (AMC) with respect to interference. When reducing the number of nearest neighbours by reducing transmit power, the interference levels decrease and spatial reuse of the time slots becomes easier. When increasing the modulation order and/or reducing the FEC rate, the number of nearest neighbours are decreased as well. However, since the required transmission power may in fact increase, the interference levels are not necessarily reduced but more probably increased and thereby limiting the possibility of spatial reuse of time slots.

In ad-hoc networks it may be advantageous to select the transmission rate for signalling of control messages such that most of the nodes receive the reservation decisions. This can be obtained by selecting a low enough transfer rate for the control signalling. The information traffic can then be transmitted at a higher transfer rate, still reaching the intended receive nodes. The initial specification of the interface between the MAC and PHY layers [26] include several measures of link quality:

- Receive data quality
- Monitor channel
- Frame errors
- Signal to noise ratio

A combined and compact metric could be developed, intended for transmission of link quality together with neighbour information utilised for selecting routes for uni- and multicast as well as selection of transmission rates.

3.5 Logical Link Control

The logical link control (LLC) layer typically provides segmentation of messages to be sent and reassembly of received messages. Priority, QoS handling of the transmit queue(s), flow and error control are handled as well.

A hybrid scheme for retransmissions is available from the physical layer. Instead of utilising pure automatic repeat request (ARQ), this hybrid combines forward error correction and ARQ. It is assumed that the hybrid ARQ scheme will be of type II, where additional parity bits are computed during the coding process, but not transmitted before the receiver sent a message to transfer additional redundant bits to enable decoding of the message. In a type I system, the request would lead to retransmission of an identical message, where the information contained in the first message is not utilised in the decoding process.

3.6 Network timing

Most of the discussed protocols require a common time reference. Civilian systems may utilise satellite navigation systems such as the U.S. based GPS, the European initiative Galileo or other available systems. Node positions may be extracted from satellite navigation as well, and it is not unrealistic that implemented radios will include satellite navigation equipment. It seems relatively easy to jam such systems, and their applicability in critical situations when jamming is involved might be questionable. It is therefore of interest to investigate the accuracy of for example crystal oscillators and distributed timing algorithms that are able to function without external satellite or terrestrial timing systems. The algorithm selected should be reasonably scalable and introduce limited extra transmissions in the time slotted network. Robustness against fading, interference and jamming attacks are of interest as well [80]. It should be noted that RBCI functionality may require a global real-time network time reference to interoperate with RBCI not requiring for example air planes to obtain local network timing.

Most of the discussed MAC algorithms would benefit from a global reference time common to all nodes, however, strictly speaking only local timing in the one hop neighbourhood is required. If the network gets clustered into groups and then are required to join together again, a stable global time synchronisation would be beneficial for the time slotted system.

The draft MAC-PHY interface specification [26] defines two parameters that the MAC layer can utilise to adjust the start of burst transmission:

- Receive acknowledge of initial acquisition, with reliability information
- Preamble Detect, with reliability information

The MAC algorithm can instruct the physical layer to advance or delay transmission with the two parameters:

- Delay instant of transmission
- Advance instant of transmission

The propagation delay is variable, and we have assumed a maximum delay of 170 μs. In addition, the multipath propagation may be severe in hilly terrain, with strong reflections arriving up to perhaps 50-100 μs after the main path. It is therefore expected that precise estimates of propagation delay from the physical layer in some cases may be difficult to extract. A measurement campaign is planned to be performed in Norway during 2008 that may provide more firm answers regarding the radio channel properties at VHF and UHF military frequencies.

A hardware clock tries to estimate the time based on for example a crystal oscillator with an approximate time. Clocks are subject to drift and variations in the oscillator frequency due to for example physical chocks and temperature variations. Crystal oscillators found in consumer electronics typically has a frequency stability of about 1 to 100 parts per million (PPM) [81]. From the MAC layer point of view, there is limited use of a true global time, and only relative time in the network is of importance. Thus, nodes can run their clocks independently and keep track of the relative drift and offset of their own clock to the other clocks in the network. Any node is then able to convert its own time estimate to any other node's time estimate.

There are two main approaches for distributed timing algorithms: physical layer based and packet based synchronisation.

Packet based

The reference broadcast protocol (RBP [81]) synchronises a set of receivers with each other. This is obtained by sending reference beacons to neighbours and time stamping the arrival times. The receivers then exchange their receiving times and estimate their relative phase offset using least square based linear regression. The method is capable of synchronising multi-hop networks with relative good precision. The beacon transmissions are not required to be additional burst, as for example RTS/CTS messages time stamped at the MAC layer can be utilised. According to [82] the algorithm has not been extended to large multi-hop networks.

An alternative protocol denoted timing sync protocol for wireless sensor networks (TPSN) was reported in [83]. It is a hierarchical approach where a root (leader) node is first selected and the other nodes then synchronise according to their level in the hierarchy. Shortcomings of the approach are the lack of time drift estimate limiting accuracy and handling of topology dynamics [82].

Physical layer based

In [89] a physical layer based distributed algorithm is described where each node continuously updates its timing based on a weighted average of timing pulses from other nodes. The weight is proportional to the received signal power from each node divided by the sum of the received power from all other nodes. This approach puts more weight on timing pulses received over for example a non-faded channel than a faded channel. Synchronisation is achieved when all the timing estimates are equal.

3.7 Discussion

Only a few of the MAC approaches described in the previous sub-sections seem appropriate for the low rate MANET of interest. The performance of time spread multiple access (TSMA) will be dependent on the connectivity degree and it will not work well when the nodes are distributed within a geographically small area.

The overhead in spatial TDMA (STDMA) grows linearly with the number of nodes in the network, hence the scalability and throughput is of some concern. Dynamic distributed TDMA enables less signalling for longer traffic streams (for example voice) compared to the soft reservation approaches. Given a somewhat longer connection setup time, the number of supported voice conversations is expected to be higher than for the soft reservation approach. A challenge might be to distribute the reservation information and include pre-emption with the same functionality as in the soft reservation approaches. Shorter data bursts should be handled in a more contention-like approach, avoiding lengthy connection setup times. Commercially available maritime ad-hoc networks such as Sub net relay and AIS are examples of networks utilising forms of dynamic TDMA.

The soft reservation approaches are scalable with respect to the number of nodes in the network. The SRMA/PA approach includes a field in the time slot utilised for pre-emption while CATA includes a field required for multi- and broadcast, both required for the MANET under study. D-PRMA introduces the concept of multiple contention attempts in a slot to improve the probability of successful reservation. The latter soft reservation approaches utilise a slot divided into one field carrying information bits and up to four slots for signalling of control messages. To obtain an acceptable overhead ratio the signalling fields are required to be substantially shorter than the data field. This is a challenge with the current physical layer.

4 Some link layer design considerations

In the previous section two interesting MAC approaches where identified: soft reservation RTS/CTS schemes and dynamic TDMA. The main real time service considered, voice, is important for designing the frame layout. In addition, the high power amplifiers and forward error correction impose lower time limits for the mini slots utilised for control and signalling information and thereby directly influencing the efficiency of the different approaches.

4.1 Minimum burst length and required guard time

The efficiency of the MAC protocols depends on the physical layer bit rate, the overhead introduced due to for example turn-around times and the packet length. An investigation of the throughput dependence on turnaround time for the CSMA protocol for various packet lengths is reported in [85]. The throughput decreases significantly with increasing turnaround time and with decreasing packet size.

Identification of a minimum burst length from a node is of interest to investigate the possibility to send short signalling messages. Traditional analogue radios with push-to-talk functionality may have power ramp up time in the order of 10's of ms, while at the other hand for example the GSM system operates with burst lengths in the order of 0.5 ms. There is a trade off between short burst lengths and power consumption that is of interest especially for battery driven terminals. As an initial design parameter we will assume a minimum burst length of 1 ms and await comments from the industry on the implementation issues.

Another time delay of interest when identifying the required guard bands is the switching time between receive and transmit. ETSI HiperLan 2 specifies a radio turn-around time of 6 μs. As a first assumption we anticipate a turn-around guard time at each side of a burst in the order of 50 μs.

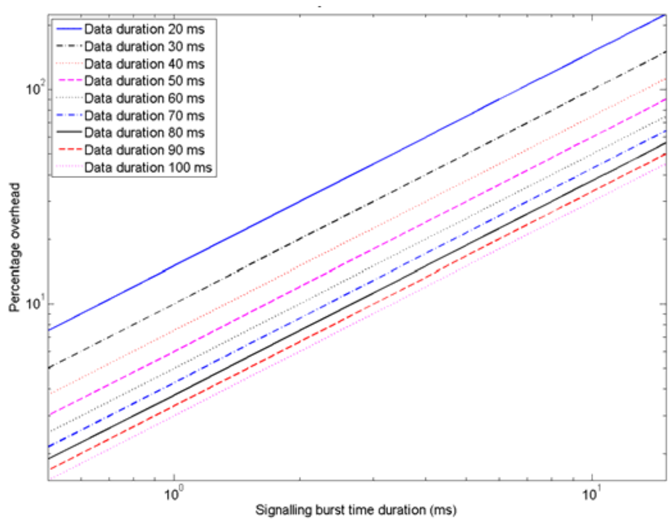
The propagation time delay will affect the necessary guard times as well. The guard time should take into account the maximum distance between nodes. If we assume this distance is 50 km, the possible transmit propagation time difference is about 170 μs when neglecting multipath propagation. A transmitted burst that require an answer then have to take into account two times the maximum propagation delay and two times the rise and fall time of the power amplifiers. We assume a guard time of at least 200 μs is sufficient as a first estimate, depending on the chosen time synchronisation method.

The minimum interleaver length for the iterative FEC is currently 10 ms at 16 kbit/s information transfer rate. This interleaver length could probably be reduced to for example 5 ms, however, the performance penalty is unknown at this stage. The required BER and PER should be taken into account when choosing the FEC code rate and interleaver lengths.

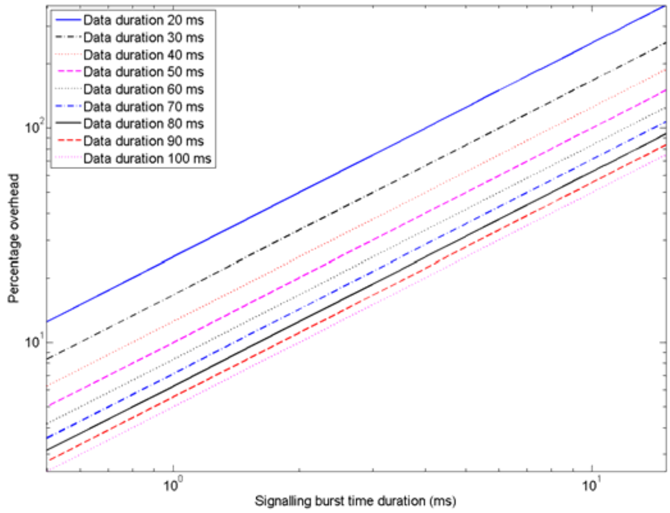
4.2 Length of MAC control messages

The efficiency of the system will to a large extend depend on the required overhead transmitted in the mini-slots. In a typical RTS/CTS dialogue, the address of the sender and the recipient should be transmitted, possibly also including priority and required QoS type. A CNR network address room will typically be fully spanned by utilising 6 bits per node (64 nodes), 8 bits would enable the maximum foreseen network size. An RTS message contains both the transmitting and receiving node's addresses, thus about 2 Bytes in total. If we assume for example 4 priority levels, 2 bits would be sufficient to handle normal/high priority for voice and data. If we allocate additional 2 bits per mini-slot for future usage, such as QoS, 3 Bytes per mini-slot might be enough. We have not yet considered the necessity of including for example initial encryption vectors in every burst. If this is required one may have to add perhaps 3-5 Bytes for every packet sent.

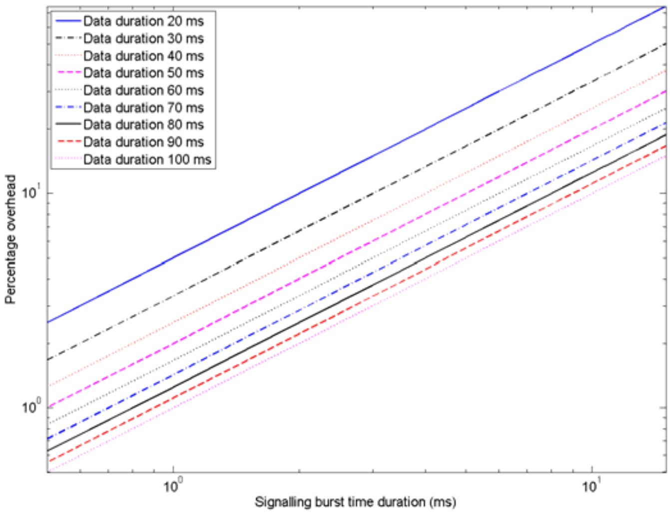
It is of interest to investigate how the overhead in terms of guard times and minimum signalling packet time duration influence the throughput on various MAC approaches. For dynamic PRMA there are at least two signalling packets for every information packet. Similarly, for SRMA/PA and CATA there are four signalling packets per information packet (including ACK for SRMA/PA). As a first estimate of the overhead we assume that the signalling packet time duration and the header in the information packet are of equal durations. This time duration then includes preamble, signalling information and guard time to combat turnaround time and propagation time delays. Thus, DTDMA contains one preamble per burst resulting in overhead. Shown in Figure 4.1 is the percentage overhead between the pure information packet excluding header and the total time duration of a packet when including delays due to signalling for the three MAC approaches.



a) Dynamic PRMA



b) SRMA/PA and CATA



c) DTDMA

Figure 4.1 Relative overhead introduced by signalling, preamble and guard time with duration of information packet as a parameter. a) D-PRMA, b) SRMA/PA and CATA and c) DTDMA

The overhead depends significantly on the duration of the signalling burst, thus it is an important design factor. The overhead depends on the time duration of the information packet as well, and long information packets impose lower overhead. For example would 10 ms signalling burst (and preamble) durations and 50 ms duration of data burst imply an overhead percentage of 100 for the CATA protocol. The corresponding overhead for the DPRMA and DTDMA protocols are 60 and 20 percent for the same durations. Shortening the signalling burst to 5 or 1 ms would reduce the CATA overhead percentage to 50 and 10 respectively. It may not be possible to utilise long packets for real time services such as voice due to time delay constraints from QoS requirements. An area of study is whether it is possible to increase the packet length by for example slot merging if allowed by the real time traffic carried by the network. This approach seems interesting when transferring larger amounts of data, such as transfer of pictures or maps.

4.3 Signalling and capacity for soft reservation MAC

In this section we provide a first iteration frame design for a modified soft reservation algorithm based on CATA, SRMA/PA and D-PRMA for a few selected bit rates. During normal operation of the algorithm, the burst (or slot) format is as shown in Figure 4.2. If there is a collision during contention, the transmitter will not receive a reservation confirmation (CTS), and instead of the data, new requests (RTSs) and possible confirmations (CTSs) are transmitted to either reserve the slot in the next frame or to send a small amount of information, if possible. The not-to-send (NTS) field is utilised during multicast reservation phase. Each transmitted burst is first preceded by a guard time, then a preamble pattern, followed by data. This is slightly different from the current physical layer specification where the parameter field is separately encoded and not combined with the data. The required information regarding for example the physical layer mode etc. is transmitted in the RTS/CTS, enabling a shorter parameter field in the data slot.

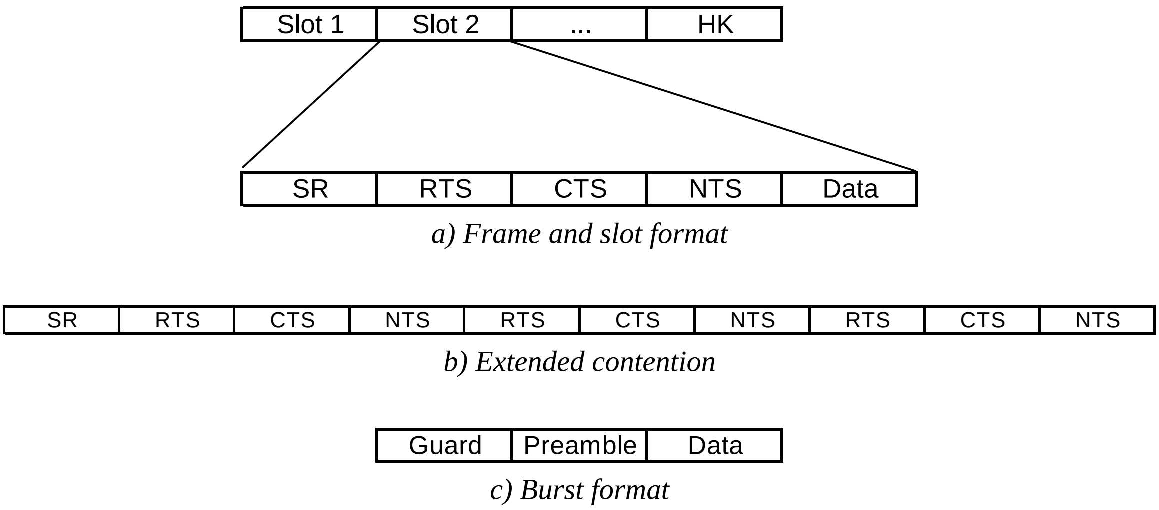


Figure 4.2 Modified slot structure. a) Frame design during normal operation. b) Operation during contention collisions during traffic flow contention process with extra contention slots for RTS/CTS/NTS. c) Burst format

As a first assumption we assume that the signalling transmission rate will be fixed for all nodes at the rates 9.6, 16, 32 or 64 kbit/s. For these cases the information bit rate is assumed identical to the signalling rate. The frames from the voice coder arrive every 22.5 ms, containing 54 bits. An integer number of the voice frames are buffered and packed into one data slot. We will assume a guard time of 500 μs, and a preamble pattern (0011...) of 1 ms for every burst transmitted, see Figure 4.2 c. Each transmitted burst contains 5 Bytes allocated for future use (for example an initial crypto synchronisation vector (IV) of 3 to 5 Bytes) and a 3 Byte parameter field. The crypto IV vector may be required if traffic flow confidentiality becomes a system requirement. In addition, the signalling fields typically carry addresses and priorities, while the data field carries user data. Thus, the signalling mini slots contain about 10 Bytes in total. The main results regarding burst durations and number of supported voice channels are given in Table 4.1. The numbers represent a first estimate and are expected to change if detailing the approach.

Transfer rate (kbit/s)	9.6	16	32	64
Guard time (ms)	0.5	0.5	0.5	0.5
Preamble duration (ms)	1	1	1	1
Bytes per signalling slot	10	10	10	10
Signalling slot duration (ms)	9.83	6.5	4.0	2.75
Data slot duration (ms)	58.8	35.9	18.7	10.1
Slot duration (ms)	98.1	61.9	34.7	21.1
Frame duration (ms)	202.5	202.5	202.5	202.5
Frames per second	4.9	4.9	4.9	4.9
Housekeeping duration per frame (ms)	6.25	16.9	29.1	12.7
Housekeeping slots per frame	0.6	2.6	7.3	4.6
Additional no. contention in data slot	2	1.8	1.6	1.2
Slots per frame	2	3	5	9
No. voice channels	2	3	5	9
Voice coder delay (ms)	43	43	43	43
Typical minimum one hop voice delay (ms)	145	145	145	145
Typical minimum two hop voice delay (ms)	345	345	345	345
Typical minimum three hop voice delay (ms)	445	445	445	445

Table 4.1 Slot format for voice transmissions

The number of voice channels supported may serve as one benchmark of the system. At 9.6 kbit/s, the number of voice channels is 2 with the given assumptions. At 16 kbit/s, 3 simultaneous voice channels are supported. For 32 and 64 kbit/s the corresponding number of voice channels is 5 and 9. The one-hop time delay excluding voice coder delay is about 200 ms. The required maximum time delay may be debated, however, one should keep in mind that at VHF the majority of push-to-talk traffic is expected to take place within a one-hop neighbourhood, where even larger time delays than 200 ms might be acceptable.

If a contention fails in the normal contention RTS/CTS/NTS dialogue, there are 1-2 extra contention attempts available in the unused data slot. This will increase the probability of successful contention somewhat, and may be used for separating low and high priority access. If there is a collision, for example only the high priority traffic is allowed to contend in the first extra slot. Another possibility is to let low priority traffic contend only in the extra slots, given that the network load is sufficient to utilise most of the slots for high priority traffic. There are a large number of possibilities available, however, utilisation of queue information would certainly be beneficial. Queue length and priority could be piggybacked to the data, or signalled in the PAR field if a sufficient number of bits are available. It should then be possible to let only high priority traffic contend and suppress low priority traffic, ensuring a reasonable contention success and hinder the system to enter into the unstable region with low throughput. Further studies are required in this area of connection admission control and network stability to identify a viable solution. One methodology that might be viable is the application of the Seedex protocol for the contention phase [86].

The frame format also allows insertion of short housekeeping slots (HK) per frame. Assuming one HK slot per frame results in about 5 mini slots per second available for transmission of for example neighbourhood information required at the routing level, position and queue updates. These slots could be allocated to all nodes in the network according to a defined schedule to distribute neighbour information, link qualities, queue lengths and the priorities. If we for example assume a terminal should have one housekeeping slot every 5 second, about 25 nodes would be supported. A larger network would then imply longer time between neighbourhood announcements and position updates. Slots not utilised could be used for short messages as a best effort short message service. Nodes that do not need to send updates in the housekeeping slot keep silent. These slots may be detected by a preamble sense scheme enabling steeling of part of the housekeeping slots. Preamble sense is one of the variables available to the MAC layer [26].

The effective bit rate for data transfer in one slot per frame without acknowledgements is identical to the design rate, 2.4 kbit/s.

Data transfer in broad- or multicast mode is expected to take place at a rate enabling mainly one-hop connections, that is, the highest rate not changing the topology from the lowest rate option. In some cases, such as unicast, the data rate for one link may be increased. The signalling is assumed to be transmitted at either 9.6 or 16 kbit/s to maintain the integrity of the reservations, while the data field may optionally utilise 32 or 64 kbit/s. Maintaining the same slot structure as for voice, it is possible to calculate the effective bit rate if a user reserves one slot per frame, see Table 4.2.

Transfer rate signalling (kbit/s)	9.6	9.6	9.6	16	16
Transfer rate data (kbit/s)	16	32	64	32	64
Effective bit rate, one slot per frame (kbit/s)	4.2	8.7	17.8	5.1	10.6

Table 4.2 Effective transfer rate for unacknowledged data transmission without slot merging, low rate signalling

The maximum effective bit rate is higher when signalling at 9.6 kbit/s compared to signalling at 16 kbit/s. This is mainly due to the longer time duration of the data slot in the former case.

In some cases an explicit acknowledgement field may be useful. In this case, the duration of the data slot has to be reduced by the time it takes to send an extra control package, that is 6.5 ms in the 16 kbit/s case and 9.83 ms in the 9.6 kbit/s case.

Transfer rate signalling (kbit/s)	9.6	9.6	9.6	9.6	16	16	16
Transfer rate data (kbit/s)	9.6	16	32	64	16	32	64
Effective bit rate, one slot per frame (kbit/s)	1.9	3.4	7.2	14.7	1.9	4.1	8.5

Table 4.3 Effective transfer rates for acknowledged data transmission without slot merging

It should be noted that ACK messages may be included in the next reserved slot, avoiding allocating part of the data field to immediately acknowledge reception.

If a node reserves two or more consecutive slots, the efficiency would be increased if the slots were merged and only one control phase was utilised. This could be signalled in the soft reservation field, ensuring that nodes not detecting any SR field(s) in the merged slots attempt to compete for the slot by sending an RTS message. An example where two slots are merged is given in Figure 4.3.

SR	RTS	CTS	NTS	Data

Figure 4.3 Slot merging example

The topology information required for the routing protocol(s) could be handled by the same slot arrangement, or super frames could be designed to send regularly differential updates of local neighbourhood and position updates say, for example, every 10 s. Full updates could be transmitted together with the two-hop neighbourhood at a lower rate.

Midambles have not been taken into account yet. These may be required in environments with long multipath delays and high data rates. The final frame format should enable insertion of one or more midambles to be used to improve synchronisation and equalize the channel if required.

4.3.1 Signalling of control messages

As discussed in for example [87] a multicast specific challenge is that some but not all of the intended receivers of a multicast message may be able to receive it due to for example interference in their neighbourhood, propagation degradations or jamming. An example topology used to provide signalling examples for uni- and multicast is shown in Figure 4.4.

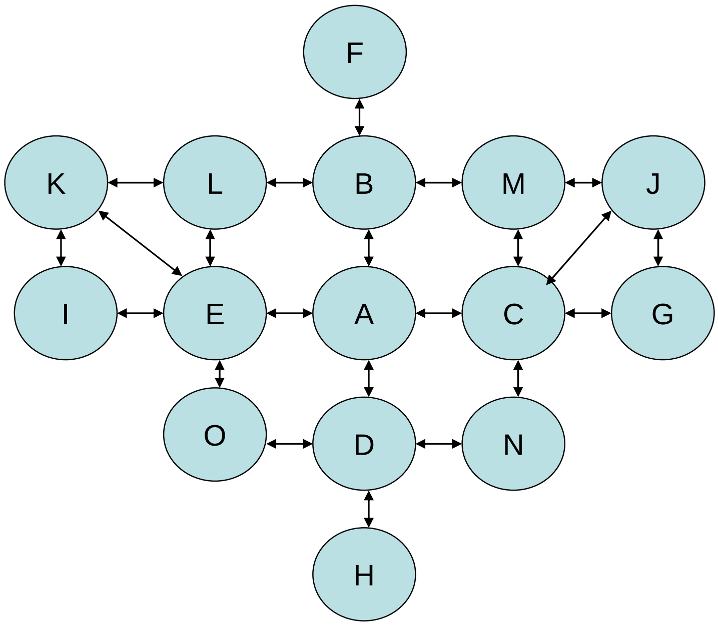


Figure 4.4 Example topology

If for example node *J* is transmitting to node *G*, node *C* is not able to hear multicast traffic from node *A*. If *A* has a high priority multicast message it should probably be able to utilise pre-emption to be able to transmit the message reliable to the nodes in its neighbourhood (*BCDE*). This is necessary as *A* may not be able to find a common available time slot where the neighbourhoods of *B*, *C*, *D* and *E* are simultaneously quiet.

For a node to be able to relay a real time multicast message, for example a transmission path through nodes *A*-*C*-*G*, the frame should contain at least two time slots. A larger number of slots would enable the traffic in the two-hop neighbourhoods of node *A* to continue in slots not reserved by *A* for multicasting.

One objective is not to relay on any carrier sense mechanisms during signalling. This is motivated by the objective of designing a robust network where jamming in the right time slots would not cause the network to break down easily. Thus, simultaneous transmissions will result in collisions where capture may occur mainly in cases where the same mini slot is utilised in two different neighbourhoods. For a signalling message to be utilised for control, it should be received and decoded correctly. Encryption at the mini slot and slot level would make it a more difficult target for intelligent jamming.

4.3.1.1 Multicast

An example control signalling dialogue for multicast is shown in Figure 4.5 where slots are represented by rows.

Multicast from A

Successful contention

SR	RTS	CTS	NTS	Data
	A	A		A
A	ABCDE			A

Unsuccessful contention for A

One NTS message

SR	RTS	CTS	NTS	Data
	G	J		G
J	G			G
J	GA	A	C	G
J	G			G

Two NTS messages

SR	RTS	CTS	NTS	Data
JK	GI			GI
JK	GIA	A	CE	A

Collision for A and B

No NTS message

SR	RTS	CTS	NTS	Data
	AB	AB		AB

Grabbing a unicast flow by A (high priority)

Neighbourhood of E

SR	RTS	CTS	NTS	Data
	I	K		I
I	I			I
I	AI	A	E	A
A	ABCDE			A

Neighbourhood of C

SR	RTS	CTS	NTS	Data
	G	J		G
J	G			G
J	GA	A	C	A
A	ABCDE			A

Grabbing a multicast flow by A (high priority)

Neighbourhood of E

SR	RTS	CTS	NTS	Data
	I	I		I
I	IKE			I
I	AIKE	A	E	A
A	ABCDE			A

Neighbourhood of C

SR	RTS	CTS	NTS	Data
	G	G		G
G	GCJ			G
G	AGCJ	A	C	A
A	ABCDE			A

Figure 4.5 Multicast signalling for a soft reservation protocol

Two or more nodes in the same neighbourhood may simultaneously start multicast reservation and thereby collide. Their neighbours will then normally not be able to respond with an NTS message and the transmitting nodes continue to send the messages. The same would apply if two nodes do not have any common neighbours. If a neighbouring node somehow is able to detect the collision, an NTS message may be sent from this neighbour. This could be in the form of for example carrier sense, or perhaps preamble sense able of detecting colliding transmissions. Both in CATA [72] and the multi channel follow up version described in [88], this problem was addressed. The chosen solution to this rather rare conflict (at least for voice transmissions) was to piggyback listings of all reserved multicast links periodically together with the data. It is not clear how two neighbouring nodes transmitting simultaneously would be able to decode the piggy backed data at the same time the node itself transmits. However, it is possible to send the listing in the housekeeping slots and thereby reschedule colliding multicasts after some time. An alternative is given in the FPRP protocol [48] utilising a form of reservation acknowledgement. We could turn the problem around and require an acknowledgement in the form of a CTS from a randomly selected neighbour before a multicast transmission is commenced. This would in some cases resolve the conflict when there are common neighbours. Nodes not hearing the busy tone from the receiver in the SR slot may then start competing for the channel, breaking the dual busy tone scheme and blocking ACKs from the receiver.

Due to collisions in case of several simultaneously transmitted NTS messages a carrier or preamble sense arrangement is required for ensuring reliable multicast reaching all of the nearest neighbours. Without sensing there is no guarantee that all of the neighbours hear a multicast transmission as the transmitting node is not able to detect the NTS message due to collisions. If retransmission is required, the retransmitting nodes have to send NTS messages if they have no available slots, or retransmit with a high enough priority.

The above discussed approaches do not guarantee delivery of multicast/broadcast information to all neighbours as temporal channel degradations or interference experienced by some nodes are not detected.

4.3.1.2 Unicast

An example control signalling dialogue for unicast is shown in Figure 4.6.

Successful contention

SR	RTS	CTS	NTS	Data
	A	B		A
B	A			A

Extended contentention

SR	RTS	CTS	NTS	RTS	CTS
	AB			A	C
C	A			A: Data	

Grabbing by C to D

SR	RTS	CTS	NTS	Data
B	A			A
B	A	C	D	C
D	C			C

Figure 4.6 Signalling for unicast for a soft reservation protocol

The unicast case is rather straight forward. If the SR mini-slot is clear, a node *A* may transmit an RTS message to the intended node. If the RTS is successfully received, the receiving node, for example *B*, replies with a CTS message and sets up a busy tone in the SR mini slot. If the CTS from *B* is correctly received by *A*, data transmission starts in the first data slot. The same time slot in the following frames is reserved until the traffic flow ends.

4.4 Signalling and capacity for distributed dynamic TDMA MAC

At a transfer rate of 9.6 kbit/s the soft reservation scheme supported two voice channels, while it may be possible to support three assuming 2.4 kbit/s voice coders with limited signalling overhead. In this section we try to reduce the number of control slots and utilise information regarding slot reservations to reduce the overhead. The main drawback of the modified soft reservation approach is the transmission of four control bursts per information burst. Especially for voice group communications the amount of control information is quite large for the relatively long holding times. Measurements of voice conversations during disaster scenarios indicate that the call holding times are lognormally distributed with mean duration significantly longer than the duration of slots in our case, see for example [89], [90], [91] and [92]. It may therefore be advantageous to spend some more time on setting up the voice reservation and utilise less resources during the PTT session. The same applies in cases with larger data transfers requiring several slots where reservation may become more effective than contention for single slots. This brings the focus towards so called dynamic TDMA with a contention phase for reservations followed by efficient transmission in reserved collision free slots for voice and long data messages. We will here pursue a distributed reservation approach without selecting cluster leaders.

The busy tone soft reservation field may be removed if information regarding the slot reservations is available to all nodes in the neighbourhood. Due to the channel time dynamics and node mobility the reservation information may have to be retransmitted on a regular basis to avoid the problem of hidden nodes. Distribution of slot reservations should be available in the two-hop neighbourhood and the receiving node should have this responsibility. It may therefore be difficult to remove the busy tone field (SR) completely, but we may allow transmission of this field at a slower rate than the data itself, for example every 2nd slot or another cyclic scheme trying to resolve colliding SRs in the two-hop neighbourhood. Pre-emption can be performed in free housekeeping or data slots. If a node in the two-hop neighbourhood understands that there are two colliding traffic flows, it could utilise pre-emption as a mean to resolve the conflicting schedules. If a mini-slot accompany every traffic slot, it may well be utilised for acknowledgement purposes when available, useful for transfer of longer TCP data streams. RBCI listening may be performed in one or more mini slots in selected frames.

The mini-slotted approach of PRMA seems to be advantageous, where we assume application of preamble sense in the mini slotted reservation scheme. To enable a reasonable chance of connection setup and transfer of data we require at least one slot per frame not reserved for voice communications, or define a super frame with several contention slots. An alternative is to utilise preamble sense with different priorities implemented in different contention periods.

The NTS field utilised for multicast can be replaced by selecting proper nodes to send CTS serial messages, assuming topology information is available. The nodes responsible for CTS responses could be selected on the basis of required retransmissions to cover the multicast group in cases where two or more hops are required. It is unsure how this would scale in a large network, and an extended NTS time slot with preamble sense enabling a better chance of at least one successfully received NTS message may be preferable. This corresponds to the PRMA contention case displayed in Figure 3.5. A third possibility for multicast reservation is to require a successful reception of a CTS message from one selected node. In the same mini slot, other nodes are allowed to transmit NTS messages if required, colliding with the CTS message and thus stopping the reservation. Simulations may be necessary to determine the most efficient approach.

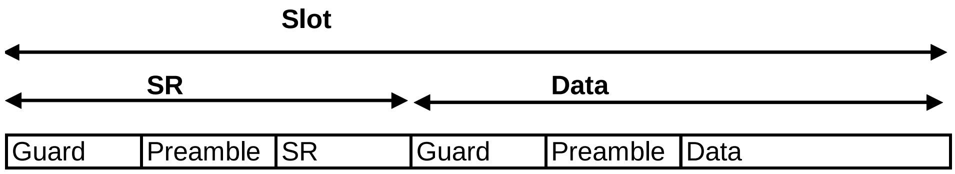
An initial frame format for 9.6 kbit/s operation is presented in Figure 4.7 assuming buffering of 10 voice frames (225 ms delay), 1 ms preamble and a guard time of 0.5 ms. It is here assumed that every data slot is paired with a slot signalling for example a busy tone with reservation information (SR field) transmitted from the intended receiver. If a lower rate of busy tone indicators is sufficient, the field may be regularly used for other purposes such as, acknowledgements or other data/control information. It is also possible to completely remove this field. Simulations are considered necessary to determine the throughput versus efficiency for the numerous variations possible.

HK	Slot 1	Slot 2	Slot 3

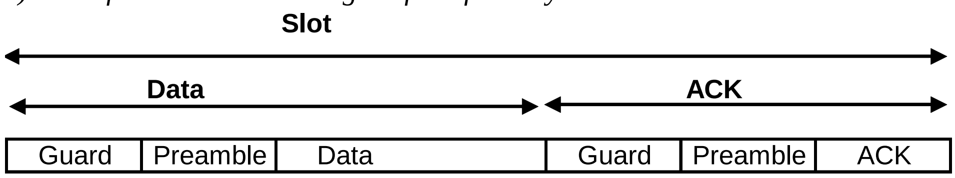
a) Frame format 1: 3 data/signalling slots and one housekeeping (mini) slot per frame

HK	MS 1	MS 2	MS 3	DS 1	DS 2	DS 3

b) Frame format 2: 3 signalling slots, 3 data slots and one housekeeping slot per frame



c) Slot format when utilising SR field for busy tone and reservation announcement



d) Slot format when utilising the mini-slot for acknowledgement

MS 1	MS 2	MS 3	MS 4	MS 5	MS 6	MS 7

e) One slot utilised for contention (mini slotted version)

HK	Frame 1	Frame 2

f) Super frame example

Figure 4.7 Frame and slot format at 9.6 kbit/s operation of dynamic TDMA

A frame consists of three slots enabling 3 simultaneous voice channels. In addition the frame includes one housekeeping slot for transmission of neighbourhood information, position updates and possibly pre-emption. Additional information may be included in a superframe structure. With an information bit rate of 16 kbit/s one possible frame layout is shown in Figure 4.8, assuming a buffering delay of 270 ms (12 voice frames).

HK	Slot 1	Slot 2	Slot 3	Slot 4	Slot 5

a) Frame format 1: five data/signalling slots and one housekeeping slots slot per frame

HK	MS 1	MS 2	MS 3	MS 4	MS 5	DS 1	DS 2	DS 3	DS 4	DS 5

b) Frame format 2: 5 signalling slots, 5 data slots and one housekeeping slot per frame

MS 1	MS 2	MS 3	MS 4	MS 5	MS 6	MS 7

c) One slot utilised for contention (mini slotted version)

HK	Frame 1	Frame 2

d) Super frame example

Figure 4.8 Frame and slot format at 16 kbit/s operation

The 16 kbit/s case would enable 5 simultaneous voice channels. The examples of dynamic TDMA shown in this section would enable one additional voice channel at 9.6 kbit/s compared to the soft reservation approach and additionally two voice channels at 16 kbit/s. This improvement is expected to be obtainable in network with slowly varying topology dynamics as expected at for example VHF frequencies. If topology dynamics increase or traffic characteristics change towards burst communications, the soft reservation approaches represent viable alternatives.

4.4.1 One hop reservations

An example multicast signalling approach is depicted in Figure 4.9. A node sends an RTS message. If semi-reliable multicast is required, it waits a number of mini slots for reception of potential NTS messages. If no NTS messages are successfully received, the node confirms the reservation with a CON message and the neighbouring nodes all transmit busy tones in the SR field to signal the occupancy of the slot in the two-hop neighbourhood. If all NTS messages collide, the node wrongfully assumes the reservation was a success, leading to a deadlock situation. When transmitting unreliable multicast, the NTS message is not used. If any of the neighbours is already active, the multicast node would have observed the transmission in the SR field.

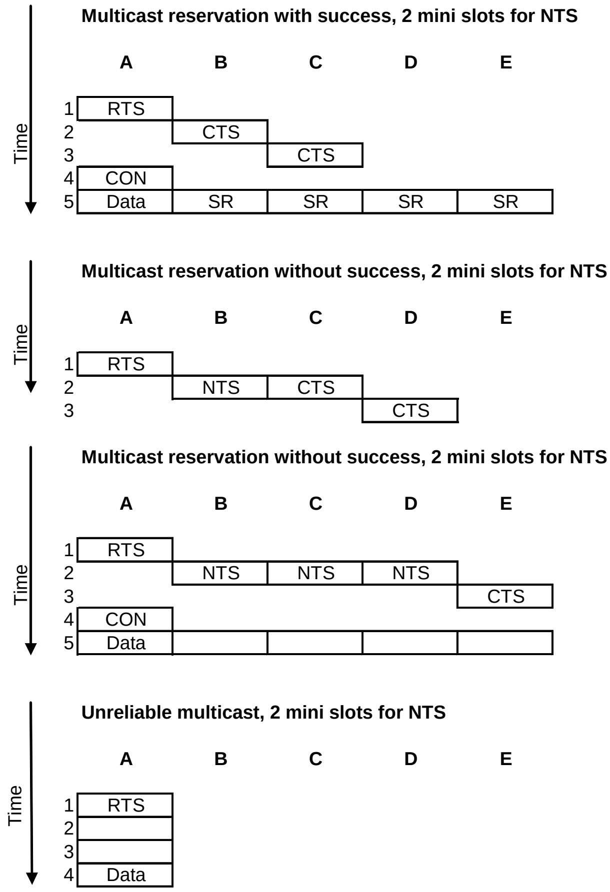


Figure 4.9 Multicast one hop- reservation signalling example for dynamic TDMA

An alternative is to require received CTS messages from a few selected nodes and allow eventual NTS messages to collide, destroying the reception of one or more CTS messages.

The unicast signalling case is shown in Figure 4.10.

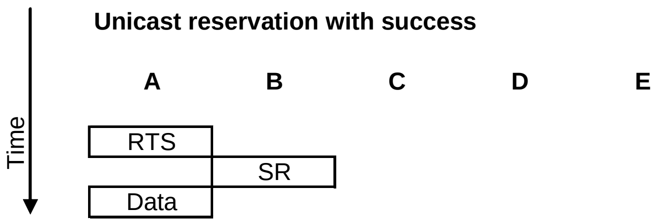


Figure 4.10 Unicast one-hop reservation signalling example for dynamic TDMA

If a node in the two-hop neighbourhood receives more than one busy tone indication in the SR field it may not be able to detect it and assume the slot is free for use. This type of problem is also present in the soft reservation approach and is difficult to avoid without utilising a carrier sense mechanism.

It may be impractical to utilise the longer connection setup routine for transfer of short data messages as termination of connections have to be announced in addition to reservations. These short messages could either be transferred in spare housekeeping slots, or with "on shot" transmissions such as slotted Aloha or preamble sense where the time slots are not reserved in the following frames. Another possibility is to utilise the soft reservation mechanism discussed earlier for shorter messages and dynamic TDMA for the longer utilising a common slot and frame format.

4.4.2 Unicast multihop voice setup

When setting up a unicast traffic flow over several hops for real time services (voice) at least two approaches are possible:

- Opportunistic signalling
- End-to-end signalling

In an opportunistic approach, a node *A* in Figure 4.11 reserves a unicast flow to node *B* and start sending data. Node *B* buffers the data and tries to reserve a unicast connection to node *C*. If successful, the voice traffic is received by *C* with an unknown delay depending on whether there are free slots available between *B* and *C*, or not. If node *B* has to perform pre-emption, additional time delays are expected.

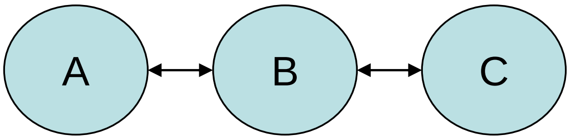


Figure 4.11 Unicast voice setup topology

An end-to-end oriented approach requires that node *A* receive an acknowledgement from both nodes *B* and *C* before starting to transmit the real time data. In this case a reservation through the MANET is obtained, that may have to be rerouted if the topology changes with accompanying signalling traffic. If we somewhat unrealistically assume that node *B* is aware of both whether there are available free slots and the priority of ongoing transfers, and whether node *C* observes the same slots as interference free, node *B* could optimistically reserve capacity to *C* and acknowledge the request from *A*. The first part of the assumption is normally fulfilled, however, the second part requires that *B* is aware of all the traffic in its two-hop neighbourhood. If *C* is occupied in either transmitting or receiving a message, node *B* would have been notified. However, if *C* is not participating in any transfers, but some of its neighbours are, node *B* has to try to allocate available time slots until it succeeds with an acknowledgement from *C*. During the time *B* investigates possibilities with *C*, *A* may receive a temporary acknowledgement from *B* to reserve the first part of the path. The procedure is illustrated in Figure 4.12.

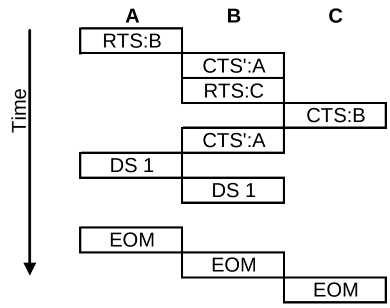


Figure 4.12 Possible voice setup signalling

As seen in the figure, at least 5 mini slots are required before *A* can start transmission of data, and the first data slot is available at *C* after 5 mini slots and two data slots. At the end of message, both nodes *A*, *B* and *C* have to signal that the time slots are free for use either explicit or implicit. The procedure may be extended relatively easily to include more hops, where up to 3 hops a new time slot has to be available. Outside the two-hop neighbourhood the slots can be reused, an additional node *D* on the right side of Figure 4.11 can reuse the slots utilised by node *A*.

4.4.3 Multicast multihop voice setup

It is assumed that the node(s) selected for the required CTS are selected as the nodes responsible for retransmission of the voice flow in an additional free slot. The selection may be based on a combined metric including connectivity, link quality, available capacity and perhaps geographical position. This is quite similar to the OLSR routing protocol where multi relay points (MRPs) are selected. For a semi-reliable multicast service the originating node is required to receive acknowledgements from these relay points, and possible children relay points. It is foreseen that setting up a semi-reliable multihop multicast real time service requires significant signalling. The degree of (un)reliability may be selected by requiring only a subset of the relay points to acknowledge successful retransmit reservations. At a minimum the originating node does not require any acknowledgements for retransmission of the voice call, resulting in a best effort type of call corresponding to the situation today with analogue radios without relays.

The topology example used is depicted in Figure 4.4. We will start with two cases, unreliable multicast and semi-reliable multicast to investigate the varying degree of signalling overhead required for voice transfer.

With *unreliable multicasting* we do not require acknowledgements from selected relay nodes nor from any of the nodes in the multicast group. In this case, no further signalling back to the originating node is required compared to the one hop unreliable multicast case described in the bottom of Figure 4.9. The selected relay points perform the same reservation attempt as the originating node, without requiring explicit confirmations. This is a form of best effort voice service that perhaps may be compared with analogue CNRs operating without relays.

To add functionality more similar to the relays/repeaters utilised in current CNRs the users would expect the relay points to actually forward the voice messages. The one hop case is given in Figure 4.9. Two hop semi-reliable multicasting then requires that the relay points confirm that they are able to reserve resources to forward the traffic. Since we have not defined any dedicated relay nodes yet, we are in this first attempt inclined to require confirmation from all of the relay points. If we later introduce special relay points operating at for example two frequencies to increase the capacity, the network coverage could be planned and the users would know in which directions the relay nodes extend the coverage. Assume node *A* selects nodes *B*, *C* and *D* as relay points. One example signalling dialogue is then represented in Figure 4.13.

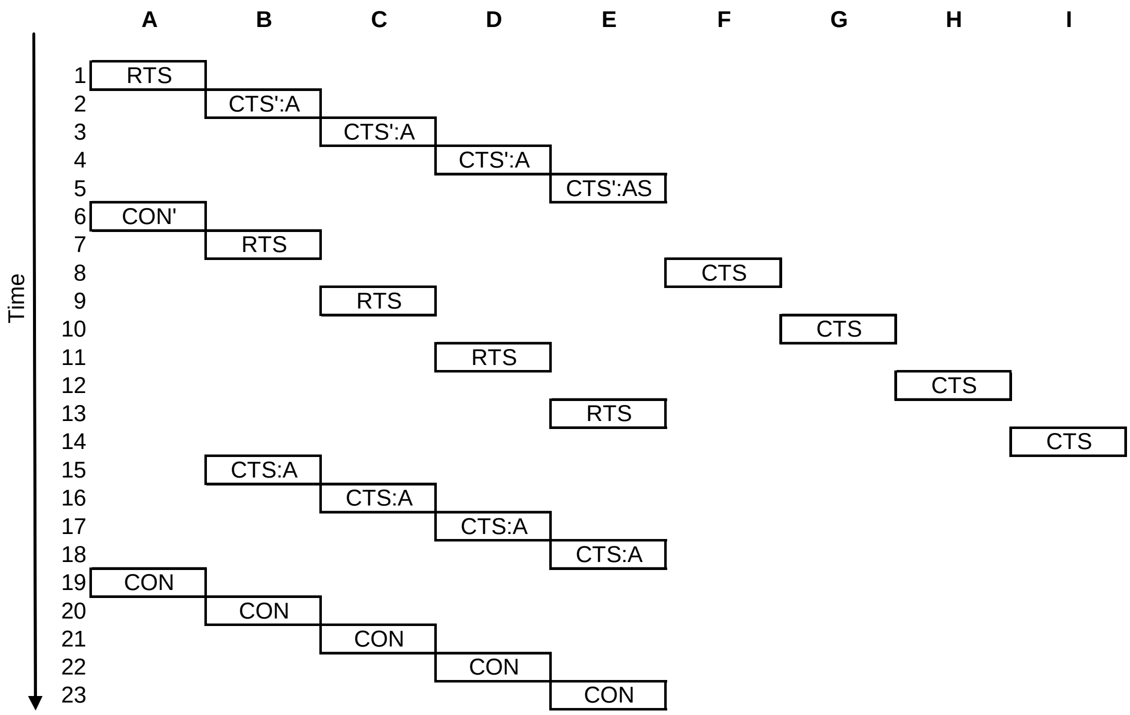


Figure 4.13 Two hop semi-reliable multicast, successful reservation

As seen in Figure 4.13, two hop semi-reliable multicast with 4 relay points would require a minimum set up time corresponding to 18 mini slots. This may be acceptable for voice communications. If reservation confirmations from node *A* and the relay points are required, additional 5 mini slots are necessary. Note that the initial requirement on voice call setup time is 250 ms.

Such a semi-reliable scheme applied in case of three hops may both exceed the required setup time and consume the majority of the resources in the network when operating at lower bit rates.

4.4.4 Non-realtime data transfer

For non-realtime data transfer it is reasonable to assume that nodes are able to buffer the received data. In such a case the number of acknowledgements may be reduced such that they represent the next hop only. If a node is not able to relay data, it may have to notify the sender depending on the application requirements. The details are left for further study.

4.5 Single node merging or leaving the network

A single node without any neighbours should transmit a beacon, for example a “hello” or RTS periodically with a slight random time variation and then listen for any confirmation CTSs from one or more terminals. If one CTS is received, a network is established and the first node should transmit an acknowledgement and start exchanging information such as addresses. If there are two new or more nodes answering with CTSs at the same time, they would detect the collision when not hearing an acknowledgement. The time for the next CTS should then be backed off randomly to try to avoid new collisions.

When a node is not heard from within a given time, to be defined, it can normally be considered as removed from the network. Radio silence should, however, be possible where nodes are able to listen but not transmit. One way of obtaining this behaviour is for nodes entering radio silence to notify its neighbours in their housekeeping slot. This information should be available at the application layer as well.

4.6 Network splitting and merging

A larger network splitting into several smaller networks is not expected to cause significant problems at the MAC layer. When two or more networks come into radio range of each other the interference levels will increase. The networks may have different timing (start of frames) and may operate at different bit rates. They may also be completely different types of networks, or hostile jamming. One approach could be to let the networks detect the significantly increased interference and let the first node detecting the problem send network information (such as technical parameters and identification) periodically and then entering into a listening phase for trying to obtain the same information from the approaching network. This nodes nearest neighbours then have to enter radio silence to reduce interference at the edges of the two networks. If a common timing and transfer rate is chosen, the impact on the network throughput when merging the networks is expected to be significantly smaller.

4.7 Inclusion of radio based combat identification

The initial requirement for the narrowband waveform is establishment of radio based combat identification within about 500 ms – 1 s. RBCI might be implemented at a fixed frequency where the nodes in the network periodically change frequency to listen for any messages from nearby airplanes. If there is proper activity at this frequency, detected by for example recognition of a known/predefined preamble, the nodes should continue communicating at the designated frequency according to an RBCI standard to be defined. RBCI activity may probably be detected by listening at the designated frequency a few tens of ms. The listening is required by all nodes, thus the interval between normal operation and RBCI operation is of importance for the reduction in the time available for normal communications. For example would 25 ms listening every 500 ms represent 5 per cent of the time spent on RBCI awareness, while increasing the interval to 1 s corresponds to 2.5 per cent of the total time. One practical approach is to set aside some time for RBCI on a frame or super frame level in the initial MAC design.

5 Conclusions

The initial requirements for a tactical military MANET have been discussed and implications on the design of the link layer for a narrowband system have been investigated. The main focus is on medium access control protocols suitable for TDMA based mobile VHF and UHF combat networks carrying both data and voice traffic with different QoS characteristics. The work is carried out as part of a NATO attempt to define a CNR network enabling standardised communications between nations employed in joint operations.

In this initial time slotted design the narrowband system is assumed to occupy 25 kHz of the spectrum, thus efficient transfer of information and reasonably low overhead ensuring scalability is required to support networks of various sizes. Support for quality of service classes, prioritisation and pre-emption is required. The access to the shared radio channel is managed by the link layer MAC protocol. Terrain obstacles, interference, jamming and potentially long distances may lead to multiple hops, and the distributed resource allocation should handle both hidden and exposed nodes in a time dynamic network topology. The MAC protocols should, in cooperation with the network layer, offer uni- and multicast as well as regular transmission of position updates of nearby friendly forces.

A literature review has been performed and interesting MAC candidates identified. Dynamic time division multiple access (D-TDMA) and soft reservation schemes such as collision avoidance time allocation (CATA) are identified as potential solutions fulfilling most of the above mentioned requirements. The main challenge for both approaches is to limit overhead due to signalling of control messages while fulfilling the requirements. We have developed an initial link layer design for two candidate approaches. Initial performance assessments are presented and the characteristics of the two alternatives are compared. For a slowly varying network topology, as experienced at VHF frequencies, dynamic TDMA seems to be the most promising candidate with respect to the available traffic capacity. With increasing mobility and operating frequency, the overhead penalty of the soft reservation approaches may overcome the drawback of dynamic TDMA, where for example reservation signalling is expected to take increasingly more of the available radio resources. Examples of possible signalling approaches are given for both uni- and multicasted traffic over one or more hops. For both approaches instability of the contention mechanism(s) during heavy load requires a form of connection admission control to ensure successful outcome of the process for start-up of new traffic flows.

Furthermore, network timing approaches are studied to some extent, and network splitting and merging is briefly discussed.

Work is still ongoing on more detailed studies of both the physical and link layer for a narrowband waveform NATO standard. Simulation studies of different approaches is expected to give firmer answers on selection of both physical layer implementations such as modulation and coding, as well on link layer implementation issues regarding medium access and logical link control. Scalability to support also a wideband version of the standard requires further studies. In the future these studies are expected to cover higher layers as well, including for example network, transport and application layers.

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Abbreviations

ACK	Acknowledgement
AF	Assured forwarding
AFD	Average fade duration
AMC	Adaptive modulation and coding
AODV	Ad-hoc on demand distance vector
AP	Alternating priorities
ARQ	Automatic repeat request
ASP	Acquisition and signalling preamble
ATM	Asynchronous transfer mode
BER	Bit error rate
BI	Busy indicator
CATA	Collision avoidance time allocation
CDF	Cumulative density function
CNR	Combat net radio
CPM	Continuous phase modulation
CRC	Cyclic redundancy check
CROMA	Collision-free receiver-oriented MAC
CSMA	Carrier sense multiple access
CTS	Clear to send
DSR	Dynamic source routing
DSRMA	Distributed slot reservation media access
D-TDMA	Dynamic time division multiple access
DYMO	Dynamic MANET on-demand
ECN	Explicit congestion notification
EF	Expedited forwarding
ETSI	European telecommunications standards institute
FEC	Forward error correction
IETF	Internet engineering task force
INSC	Interoperable networks for secure communications
IP	Internet protocol
ISI	Intersymbol interference
ITU	International telecommunication union
LCR	Level crossing rate
LDPC	Low density parity check
LLC	Logical link control
MAC	Medium access control
MACA	Multiple access with collision avoidance
MANET	Mobile ad-hoc network
MELPe	Enhanced mixed-excitation linear predictive
MLPP	Multi-level precedence and pre-emption
MOS	Mean opinion score
MRP	Multi relay points
MSAP	Mini-slotted alternating priorities
NBWF	Narrow band waveform
NC3A	NATO consultation, command and control agency
NTS	Not-to-send
OFDM	Orthogonal frequency division multiplexing
OLSR	Optimized link state routing protocol
PAR	Parameter
PDU	Protocol data unit

PER	Packet error rate
PESQ	Perceptual evaluation of speech quality
PDF	Probability density function
PPM	Parts per million
PRMA	Packet reservation multiple access
PRS	Proper robust scheduling
PTT	Push-to-talk
QAM	Quadrature amplitude modulation
QoE	Quality of experience
QoS	Quality of service
RBCI	Radio based combat identification
RBP	Reference broadcast protocol
RC	Reservation confirmation
RFC	Request for comments
RO	Random order
RR	Reservation request
RR	Round robin
RTS	Request to send
SINR	Signal-to-interference-plus-noise ratio
SLA	Service level agreement
SNR	Signal-to-noise ratio
SOTDMA	Self organizing time division multiple access
SR	Slot reservation
SRMA/PA	Soft reservation multiple access with priority assignment
STANAG	Standardization agreement
STDMA	Spatial time division multiple access
TBRPF	Topology dissemination based on reverse-path forwarding
TDMA	Time division multiple access
TPSN	Timing sync protocol for wireless sensor networks
TSMA	Time spread multiple access
UHF	Ultra high frequency
VHF	Very high frequency
WBWF	Wide band waveform