Amateur Radio Digital Voice Communications

Time's up.

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The telecommunication industry has been busy developing digital voice transmission technologies. While amateurs have made progress in other areas, digital voice has not kept pace. Maybe its because of lack of suitable standards, the unavailability of hardware, satisfaction with the existing SSB and FM voice systems or the lack of urgency.

Well, *time's* up. We need to apply the same energies and talents we did previously in developing single sideband, amateur television, small satellites and packet radio.

A design engineer once told me, the longer you wait, the easier the job. Obviously, he meant that new devices are constantly becoming available, maybe even the specific application you have in mind. This is true in the case of digital voice coder-encoders (CODECs). Instead of developing coding algorithms and designing application-specific integrated circuits (ASICs), someone else may have handled these chores for you.

CD-quality Digital Voice

Digital audio broadcasting (DAB) standards are now shaking. The original hope was that one standard could be used for both satellite and terrestrial delivery to cars, portables and the home. Eureka 147 is a system that has gained some favor in Europe and Canada. For a while, it looked like the United States was going to accept it as a standard but domestic broadcasters felt it important to keep broadcasting local and hang onto their existing market areas. This made the so-called in-band on-channel (IBOC) scheme look more attractive. In fact, it was found that stereo digital sound could be transmitted some dB down from an existing FM program in a VHF channel.

While interesting, these technologies haven't been directly applicable to amateur communications. DAB strives for high fidelity stereo, comparable to that of a compact disk. So far, at least, amateurs haven't needed CD quality. That's a dangerous statement to make, of course, as it's possible that another speaker will be making a presentation along those lines.

Digital Cellular Telephones

Here's something a little closer to our needs. Audio quality requirements are similar to the public switched telephone network. While not being CD quality, it's good enough to accommodate speech of all languages and unfamiliar topics with clarity and low noise. Amateurs can tolerate lower voice quality and lower signal-to-noise (S/N) ratios, or have conditioned themselves to it so far. Lurking there is operator pride in being able to pull a signal out of the muck. Nevertheless, just listen to some of the lousy SSB voice signals we endure on HF or even on amateur satellites. VHF/UHF FM sounds lots better. Even that isn't up to the quality of the digital cellular telephone systems.

Chips Abound

If we start with the premise that we don't need digital cellular telephone quality or don't want to devote the bandwidth and power (or S/N) they do, we still can benefit from the abundant integrated circuits spawned by this industry. A romp through the trade journals, data sheets and appnotes will boggle the mind. One way to get a crash course is to sign up for itinerant seminars put on by semiconductor manufacturers. They'll also tell you about their products for cordless phones and other wireless applications.

Communications-Quality Digital Voice

FM two-way radios are generally compatible with each other, except perhaps in channel width. Channels ratcheted down from 60 kHz in the Jurassic FM age to 12.5 kHz these days with a strong feeling that half that would be needed to *refarm* the two-way commercial spectrum. Regrettably, the quality degrades to that of AM when an AM bandwidth is reached, according to the FM Improvement Ratio curves. Both fidelity and frequency reuse are affected, so you can't expect to get a 2: 1 improvement in spectrum efficiency by chopping 12.5-kHz channels into twice as many 6.25 kHz wide.

What industry or the standards bodies have not done is to agree on a single digital voice technology for VHF/UHF two-way radios. While there is no clear winner at this stage, a number of digital voice coding schemes have been developed, notably APCO Project 25 in the United States, TETRA in Europe, IDRA in Japan and DIMRS in North America. A complete technical description of these technologies would be impractical. Fortunately, however, International Telecommunication Union Radiocommunication Working Party 8A has captured their salient features in a scant 47 pages, in a Draft New Recommendation entitled, Spectrum Efficient Digital Land Mobile Systems for Dispatch Traffic, which is reprinted with permission of the ITU as an appendix to this paper.

Of the above technologies, Project 25 looks particularly interesting in that RF channel spacings of 12.5 kHz and 6.25 kHz are shown, while the others use 25-kHz channels. Without considering other factors such as frequency reuse, it is not clear whether Project 25's reduced bandwidth offers commensurate improvement in spectrum efficiency.

There is a new US Federal Standard for a 2.4-kbit/s digital voice technique known as Mixed Excitation Linear Prediction (MELP). A. brief description can be obtained as an IEEE reprint, 0-7803-3 192-3/96 $5.00© 1996 IEEE. Listener tests have shown MELP at 2.4 kbit/s performs as well as a higher bit rate CELP at 4.8 kbit/s specified in Federal Standard 1016.

Lead, Follow or Get Out of the Way

In the case of VHF/UHF amateur digital voice communications, it looks like we ought to *follow* what industry is doing, or at least a subset thereof. After all, there isn't that much difference between our two-way radios and commercial two-way radios. Before the amateur radio manufacturers will produce digital voice products, they'll need to have confidence that the standard they use will be stable and preferably the same one as they use commercially. That doesn't translate to just sitting and waiting for the commercial shakeout. We need to do some standards shopping.

We may be able to *lead* at HF. But we must hurry. There is work going on in the HF communications and broadcasting industries. 'The communications applications of digital voice are probably easier to implement than those for broadcasting, for the simple reason that broadcasting requires cheap receivers for the mass audiences. It doesn't mean that digital communications receivers should be expensive, but there is more price flexibility and the quantities are manageable.

Is it feasible to use something like 2.4-kbit/s MELP for amateur HF digital voice? Does it need a clear channel or can it tolerate QRM? What if the QRM is on-channel or mistuned? Will it have advantages over SSB? How do we socialize the introduction of digital voice into what other hams consider SSB voice bands? Remember the onslaught of SSB into what were AM voice bands? Add more questions of your choice.

It's not generally known that the emission designators for digital voice are already written into Part 97 of the FCC's Rules. See §97.3(c)(5) and §97.305(c) Emission Types Authorized column where "phone" is permitted. At least we don't need to go though the process of getting a Special Temporary Authority (STA) to experiment with digital voice. But we do need to give thought to the socialization issues mentioned above.

Spread Spectrum and Digital Voice

We need to be sensitive to voices that say, "Take it easy; don't QRM other users in the bands while introducing spread spectrum technology." Spread spectrum is still a controversial topic among amateurs and telecommunications professionals. Nevertheless, one of its forms, code division multiple access (CDMA) is making gains commercially. SS is neither a panacea or the end of the universe as we know it and everything we hold dear. It has its applications; we just need to chose the right ones and get on with it.

Present FCC Rules permit SS on amateur bands above 420 MHz. A current STA permits SS on all bands above 50 MHz. There are those who feel that SS should be permitted under the Rules above 50 MHz and there are others who think it ought to be allowed on all bands. Maybe this debate will be settled sometime soon.

Meanwhile, however, digital voice and SS are natural companions. We need to consider whether digital voice should use SS access techniques above 420 MHz where it's already authorized. While not wishing to debate the rule-change controversy, we can still treat it like dreams of the celibate. Would digital voice on SS be the solution to the short-range/long-range QRM problem at 50 MHz? Is SS on an overlay basis compatible with existing narrow-band modes in the HF bands?

Maybe the government has HF SS systems that they'd have to kill us if they told us about them. Industry is now working on HF SS technologies but there are still many unanswered questions. But it does offer the possibility of mitigating distortion resulting from multipath fading and it could reduce interference in crowded bands given the right techniques.

What Can a Body Do?

We should be following the industry and standards organizations in their pursuit of VHF/UHF two-way radio standards. This should also include a dialog between amateurs and Amateur Radio industry. There'll be a time to move, and we should recognize it when we see it. There is room for experimentation, but this may be more of an engineering problem of scoping what we want to do and then doing it.

Digital voice for the MF and HF bands requires a different strategy. We need to study the work being done by others, come up with some ideas for amateur applications and design some experiments. Because of the vagaries of ionospheric propagation, coding schemes that work fine at VHF or even in the MF broadcast band may flake out at HF. Here is a technical area where amateurs can contribute to the radio art.

Further Reading in the Amateur Literature

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Working Party 8A

DRAFT NEW RECOMMENDATION ITU-R M.[8A/XB]

SPECTRUM EFFICIENT DIGITAL LAND MOBILE SYSTEMS FOR DISPATCH TRAFFIC

(Question ITU-R 3 7/8)

Summary

Demand in the land mobile service is on the increase due to annual growth as well as to new data-based service requirements. This has led to the development of more spectrally efficient technologies utilizing digital modulation and in many cases trunking. These technologies are being introduced worldwide to accommodate this demand.

This Recommendation provides the technical and operational characteristics for spectrum efficient digital dispatch systems for international and regional use. The Recommendation also provides details of systems being introduced throughout the world.

The ITU Radiocommunication Assembly,

considering

a) that the rapid development of digital speech transmission now enables both public and private mobile systems to operate in both national and international arenas;

b) that digital transmission techniques offer flexibility for maintaining good communications quality;

c) that equipment using digital modulation, speech coding, channel coding and digital signal processing techniques are now comparable in price with those using analogue technology;

d) that digital transmission techniques can incorporate both speech and data facilities;

e) that digital transmission systems can accommodate different types of data services;

f) that digital systems can offer greater spectrum efficiency than existing analogue systems;

g) that automatic sharing of channels using trunking is shown to improve channel efficiency;

h) that there is continuing demand for privacy facilities;

j) Question ITU-R 37/8 on systems with improved spectrum efficiency;

k) that considerable development of multichannel digital systems, worldwide, has produced regional and national standards;

l) standardization of systems is required to permit roaming;

m) that digital systems may need to be optimized for the frequency bands in which they operate;

recommends

that the following technical and operational characteristics be adopted for spectrum efficient digital land mobile systems for international or regional use:

1 General objectives

The general objectives of a spectrum efficient digital land mobile system, for dispatch in either private or public systems, are to provide:

- systems that offer a higher spectrum efficiency, thereby accommodating more users within limited spectrum resources than analogue systems;
- a higher average level of voice quality over the network and enciphered speech for privacy;
- users with a wide range of services and facilities, both voice and non-voice, that are compatible with those offered by the public fixed networks (PSTN, PDN, ISDN, etc.);
- users with a variety of applications to satisfy their requirements, ranging from handheld stations to vehicle mounted stations, with voice and data interfaces;
- mobile and infrastructure equipment which use state of the art technology to provide savings in weight, power consumption and cost.

2 Service types

The basic services offered by a digital dispatch traffic system can be divided into three types:

- teleservices;
- bearer services; and
- supplementary services.

2.1 Teleservices

Teleservices provide the user with full capability, including terminal equipment functions, to communicate with other users. These services are typified by both lower layer (OSI Layers 1 through 3) and higher layer (OSI Layers 4-7) functionality.

Typical teleservices should include:

- a trunked and non-trunked capability to permit direct mobile-to-mobile and group speech call facilities with user options to permit selective and secure calling;
- telephony, facsimile and some extended service offerings, e.g. videotex, telex, etc.

2.2 Bearer services

Bearer services give the user the capacity needed to transmit appropriate signals between certain access points These services are typified by lower layer functionality, typically limited to OSI Layers 1 through 3.

Typical bearer services should include:

- a circuit mode data facility to permit a minimum of 7.2 kbit/s for unprotected data and a minimum of 4.8 kbit/s for protected data;
- a packet mode connection-oriented data and connectionless data facility.

2.3 Supplementary services

The range of supplementary services varies depending on the system and also the particular implementation.

3 Channel design

The systems may have two types of channel categories:

- traffic channels which are used for voice and data transmission; and
- control channels which are used for signalling and control purpose, e.g. access control, broadcast messages, synchronization, etc.

4 Channel access techniques

Systems should use either FDMA, TDMA, CDMA, or hybrids of these. Digital cellular technology may be adaptable for dispatch use.

5 Systems being installed or planned

General details of the systems are given in Annex 1.

Annexes 2, 3, 4 and 5 give general descriptions of specific systems.

ANNEX1

Systems being installed and planned

1 Introduction

Digital land mobile radio systems for dispatch and fleet management applications are being developed worldwide. Although these systems have been developed to meet the requirements of either general purpose applications or more specific groups of users, they share some of the basic objectives and characteristics outlined in the Recommendation.

A description of the systems is given below and more detailed descriptions can be found in Annexes 2, 3, 4 and 5.

1.1 TETRA

The development of the Standards for TETRA has been carried out in the European Telecommunications Standards Institute (ETSI), a recognized standardization organization.

The technical requirements specification aims to satisfy the needs of a wide range of professional users ranging from emergency services to commercial and industrial organizations.

1.2 Project 25

The development of the standards for Project 25 has been carried out by Project 25, a combined effort of US local (Association of Public-safety Communications Officials International, APCO), state (National Association of State Telecommunications Directors, NASTD) and federal government users; in collaboration with the Telecommunications Industry Association, a recognized standardization organization.

The Project 25 standards aim to satisfy the needs of a wide range of users, primarily in the area of public safety and governmental operations.

1.3 Integrated Dispatch Radio System (IDRA)

The development of the Standards for the IDRA System has been carried out by the Association of Radio Industries and Businesses (ARIB) in Japan. ARIB is an external MPT (Ministry of Post and Telecommunication) affiliate, a recognized standardization organization.

The technical requirements of the specification aim to satisfy the needs of users over a wide range of professions, from emergency services to commercial and industrial organizations.

1.4 Digital Integrated Mobile Radio System (DIMRS)

The Digital Integrated Mobile Radio System (DIMRS) is one of the methods being used in North America to provide integrated dispatch services and increase spectrum efficiency.

2 Explanation of table

Table 1 presents the core parameters for these systems. In each case, complete specifications are, or will be, available from the relevant authorities as indicated in the annexes.

TABLE 1

Core parameters

Parameter	Project 25	TETRA	IDRA	DIMRS
Designation of emission				
- Traffic channels	8K10F1E, 5K76G1E(1)	25K0D7W/25KWDW(2)	20K0D7W/20KWDW(2)	20K0D7W/20KWDW(2)
- Control channels	8K10F1E, 5K76G1E(1)	25K0D7W/25KWDW(2)	20K0D7W/20KWDW(2)	20K0D7W/20KWDW(2)
Frequency bands (MHz)	Not yet determined but likely to be 130 - 200 (VHF-150) 360 - 512 (UHF-400) 800 - 941 (UHF-800)	Not yet determined but likely to be 380 - 390/390 - 400 or 410 - 420/420 - 430 or 450 - 460/460 - 470 or 870 - 888/915 - 933	Presently used 1 453 - 1 477/1 501 - 1 525 Future use 905 - 915/850 - 860	806 - 821/851 - 866
Duplex separation (MHz)	varies or none at VHF-150 band 3 and 5 at UHF-400 band 39 and 45 at UHF-800 band	5 - 10 (400 MHz band) 10 - 45 (800/900 MHz band) dependent on system design	48 (55 in the 800 MHz band)	45
RF carrier spacing (kHz)	12.5 for 8K10F1E(C4FM) 6.25 for K76G1E(CQPSK)	25	25	25

TABLE 1 (CONT'D)

Core parameters

Parameter	Project 25	TETRA	IDRA	DIMRS
Maximum base station ERP (W) - Peak - Average	 500 500	 25 25	 Not specified typically 40-300	 Not specified 250
Nominal mobile stations transmit power (W) Ileak/Average - Mobile - Handheld	 ranges from 10/10 to 1101110 ranges from 1/1 to 5/5	 10/2.5 1/0.25	 -/2 Not yet specified	 10.4/0.5 3.5/0.17
Cell radius&n) - Handheld/Suburban - Mobile/Rural	7.6 - 35 7.6 35	3.8 - 17.5 3.8 17.5	Not yet determined Not yet determined 20 - 40	5 - 40 (Design dependent) 5 40
Area coverage technique	Cellular channel reuse Simulcast Voting receivers	Cellular channel reuse Quasi synchronous (Simulcast) Time sharing transmission Diversity receivers	Cellular channel reuse Diversity receivers (Base)	Cellular channel reuse Diversity receivers
Access method	FDMA	TDMA	TDMA	TDMA
Traffic channels/RF carrier - Initial - Design capability	 1 1	 4 8	 6 6, 3, 12	 6 6, 4, 3, 8, 12, etc.
Transmission rate (kbit/s)	9.6	36	64	64

TABLE 1 (CONT'D)

Core parameters

Parameter	**Project 25**	**TETRA**	**IDRA**	**DIMRS**
Modulation	QPSK - c family includes C4FM and CQPSK	π/4 DQPSK	M16QAM (M = 4)	M16QAM (M = 4)
Traffic channel structure				
- basic rate speech codec			Bit rate with error	
- bit rate (kbitls)	4.4	4.567	protection is less than	4.2
- error protection	2.8	2.633	7.466	3.177
- coding algorithm	IMBE	ACELP	Not specified	VSELP (6:1)
Alternative rate speech codec	N/A	rate tbd	N/A	
- bit rate (kbitls)				8.0
- error protection				6.7
- coding algorithm				VSELP (3:1)
- circuit mode data				
- protected (kbit/s)	6.1	up to 19.2	up to 4.8/slot	7.2
- non-protected (kbit/s)	9.6	up to 28.8	7.466/slot	None
- Packet mode data	IP - Internet protocol	Connection-oriented connectionless	Connection-oriented (option) connectionless	Connection-oriented, connectionless Supports IP and other network protocols
Control channel structure*				- slot information channel: 1
- common control channel	-2	-2	-1	- primary control channel: 3
- associated control channel	-3	-3	-2	- temporary control channel: 1
- broadcast control channel	-2	-2	-1	- dedicated control channel: 1
* number of channel types			(option: 5)	- associated control channel: 1

TABLE 1 (CONT'D)

Core parameters

Parameter	Project 25	TETRA	IDRA	DIMRS
Delay spread equalization capability(microseconds)(3)	Class A - 50 µs Class Q - 50 µs	Class A - no equalization Class B - 55.5 µs Class Q - 111.1 µs	Class A - no equalization Class B - no equalization Class Q - N/A	Class A - 39.8 µs without equalizer Class B - 65.5 µs without equalizer Class Q - N/A
Channel coding	BCH code for network ID Trellis codes for data Golay & Hamming codes for voice Reed-Solomon codes for embedded signals	Convolutional codes with interleaving plus error detection	Multirate trellis coding with interleaving plus error detection and bit prioritization/Convolutional codes with interleaving plus error detection	Multirate trellis coding with interleaving plus error detection and bit prioritization
Encipherment – Security levels	Types 1, 2, 3 and 4	Air interface is exportable plus authentification. Plus end-to-end encryption user definable up to the highest level of security	Not specified	Allowed for
– Multi-algorithm	Yes	Yes		
– Multikey	Yes	Yes		
– Encipherment control	Yes	Yes		
– Over the air rekeying	Yes	Yes		
Handover	Yes	Yes	Option	Yes

TABLE 1 (CONT'D)

Core parameters

Parameter	Project 25	TETRA	IDRA	DIMRS
Inter-system roaming capability	Yes	Yes	Yes	Yes
Design capability for multiple operators (systems) in same area	Yes	Yes	Yes	Yes
Direct mode	Mobile-to-mobile Channel scan(4) Repeater Trunking node gateway	Mobile-to-mobile Dual watch(5) Repeater Trunking mode gateway	Not yet determined	Allowed for

NOTE 1 – Denotes the emission classifications for C4FM and CQPSK modulations. Both alternatives utilize a common receiver and are thus interoperable.

NOTE 2 – Denotes the emission classification for base stations/mobiles (handportables).

NOTE 3 – Classes A and B refer to single transmitter operation. Class Q refers to quasi-synchronous (simulcast) operation..

NOTE 4 – Scanning channels for the purpose of alternative channel communication.

NOTE 5 – Allows a terminal using Direct Mode service to monitor the trunking control channel for any incoming signalling. It also allows a terminal in trunking mode to monitor a direct mode channel.

ANNEX2

General description of the TETRA system

1 Introduction

TETRA is a high-performance mobile radio system which has been developed primarily for professional users such as the emergency services and public transport. The TETRA suite of mobile radio specifications provide a comprehensive radio capability encompassing trunked, non-trunked and direct mobile-to-mobile communication with a range of facilities including voice, circuit mode data, short data messages and packet mode services. TETRA supports an especially wide range of supplementary services, many of which are exclusive to TETRA.

TETRA is designed to operate in the bands below 1 GHz and the 25 kHz channel structure allows it to fit easily into existing PMR frequency bands.

The specifications cover three distinct telecommunication services corresponding to:

- voice plus data;
- packet data optimized; and
- direct mode.

The packet data optimized (PDO) standard is based on the same physical radio platform as the TETRA25 voice plus data standard but implementations are not expected to interoperate at the physical layer. Full interoperability is foreseen at OSI Layer 3.

Direct mode (DM) provides direct mobile-to-mobile communications when outside the coverage of the network or can be used as a secure communications channel within the network coverage area. It will interoperate with TETRA25 both at OSI Layer 1 and OSI Layer 3.

2 Services

2.1 Teleservices

Clear speech or enciphered speech in each of the following:

- individual call (point-to-point);
- group call (point-to-multipoint);
- acknowledged group call;
- broadcast call (point-to-multipoint one way).

2.2 Bearer services.

Individual call, group call, acknowledged group call, broadcast call for each of the following:

Circuit mode unprotected data 7.2, 14.4, 21.6, 28.8 kbits/sec.

Circuit mode protected data (low) 4.8, 9.6, 14.4, 19.2 kbits/sec.

- Circuit mode protected data (high) 2.4, 4.8, 7.2, 9.6 kbits/sec.
- Packet connection-oriented data.
- Packet connectionless data.

2.3 Supplementary services supported

PMR type supplementary services

Access priority, pre-emptive priority, priority call

Include call, transfer of control, late entry.

Calls authorized by dispatcher, ambience listening, discreet listening.

Area selection.

Short number addressing.

Talking party identification.

Dynamic group number assignment.

Telephone type supplementary services

List search call.

Call forwarding - unconditional/busy/no reply/not reachable.

Call barring - incoming/outgoing calls.

Call report.

Call waiting.

Call hold.

Calling/connected line identity presentation.

Calling/connected line identify restriction.

Call completion to busy subscriber/on no reply.

Advice of charge.

Call retention.

2.4 Security aspects

The TETRA system is designed to ensure high levels of security. The security objectives are listed below:

Correct charging	–	primarily of interest to commercial systems.
Authenticity	–	proving the true identity of the communicating parties and of the network.
Confidentiality of communication	–	protection against unauthorized reading of transmitted information.

Integrity of communication	–	protection against unauthorized modification of transmitted information.
Privacy	–	privacy of people using or operating the network, e.g. personal information, identities, location.
Traffic flow confidentiality	–	to prevent disclosure of information which can be inferred from observing traffic patterns.
Monitoring	–	to permit authorized monitoring of communications, uninhibited by the security mechanisms.
Security management	–	to enable administration of a secure network.

3 Overview of the system

The functional architectures for V+D and PDO are shown in Figures 1 and 2, including their respective standardized interfaces.

4 System specifications

Refer to Table 1.

4.1 Logical channels

The following logical channels are defined:

Common Control Channel (CCH) comprising:

Main Control Channel (MCCH).

Extended Control Channel (ECCH).

These channels deal with control information addressed to or received from MSs not involved in a circuit mode call.

– Associated Control Channel (ACCH) comprising:

Fast Associated Control Channel (FACCH).

Stealing Channel (STCH).

Slow Associated Control Channel (SACCH).

These channels deal with control information intended for or received from MSs involved in a circuit mode call.

Broadcast Common Control Channel (BCCH) comprising:

Broadcast Synchronization Channel (BSCH).

Broadcast Network Channel (BNCH).

These channels carry the downlink system broadcast information.

Traffic channels (TCH) comprising:

Speech traffic channel (TCH/S).

Speech or data traffic channels (TCW7.2, TCH/4.8, TCW2.4).

These channels carry the circuit mode voice or data traffic information.

4.2 TDMA frame structure – Voice and data

The TETRA frame structure, shown in Figure **3,** has four slots per TDMA frame. This is further organized as 18 TDMA frames per multiframe of which one frame per multiframe is always used for control signalling. This eighteenth frame is called the control frame and provides the basis of the slow associated control channel (SACCH).

The circuit mode voice or data operation traffic from an 1 S-frame multiframe length of time is compressed and conveyed within 17 TDMA frames, thus allowing the eighteenth frame to be used to control signalling without interrupting the flow of data. Besides the basic TDMA frame structure described above, there is a hyperframe imposed above the multiframe structure. This is for long repeat frame purposes such as encipherment synchronization. Furthermore, it can be seen that each time-slot is of 5 10 modulation bits in duration.

4.3 Burst structure – PDO

The PDO access schemes are Statistical Multiplexing (STM) for the downlink and Statistical Multiple Access (STMA) for the uplink The carrier separation is 25 kHz.

The basic radio resources are subbursts, transmitting information at a modulating rate of 36 kbit/s. On the uplink there are four types of subbursts. On the downlink, there are two types of subbursts. Figure 4 describes the PDO up and down burst format.

4.4 Traffic channels

4.4.1 Speech traffic channels

The speech codec, and the associated error correction and detection mechanisms have been defined in the TETRA standard. Speech frames of **30** ms, each comprising 137 bits provide a net bit rate of **4.567** kbit/s. The coding method, ACELP, has been designed to achieve robustness to transmission errors, and to offer a high quality in the presence of background acoustic noise while using a limited bit rate.

Error correction (consisting of a 1/3 rate punctured convolutional code) and interleaving schemes, to selectively protect the most important bits within the speech frame, have been specified. Furthermore, an error detection mechanism has been included and bad frame replacement techniques can be used, in order to minimize the impairment of the speech quality resulting from speech frames not correctly received.

4.4.2 Data traffic channels

Data services of up to 19.2 kbit/s are supported with channel coding and interleaving schemes by using up to four time-slots per TDMA frame.

Unprotected digital bearer services with a bit rate up to 28.8 kbit/s are also supported.

5 Operational characteristics

5.1 Location updating and roaming

The mobile station evaluates the received signal and initiates the location updating procedure when necessary.

A location area is the area in which a mobile terminal cap move freely without updating the location information maintained in the network. The paging area is the area in which a mobile is paged.

The Switching and Management Infrastructure (SWMI) will page the mobile terminal in every location area where it is registered.

To facilitate mobility management, a mobile terminal may be temporarily registered in a number of location areas so that a mobile terminal may travel freely between the areas without the need to reregister.

Roaming is possible within a TETRA network and between TETRA networks.

5.2 Communication protocols

The communication protocols are layered according to the OSI model and are specified in the TETRA standards.

Layers 1 to 3 are subdivided as shown in Figure 5. The C-plane corresponds to all signalling information, both control and data and also packet mode data traffic. U-Plane information corresponds to circuit mode voice or circuit mode data.

The MM, CMCE and PD are defined in Figure 5.

The MLE (Mobile/base Link Control Entity) performs management of the mobile-to-base/base-to-mobile connection, mobility within a registration area, identity management, quality of service selection, protocol discrimination (i.e., routing to the higher layer applications).

The LLC (Logical Link Control) layer is responsible for scheduling data transmission and retransmissions, segmentation/reassembly, logical link handling.

The MAC (Medium Access Control) layer performs frame synchronization, interleaving/ de-interleaving channel coding, random access procedures, fragmentation/reassociation and BER and MER measurements for control purposes.

5.3 Call set-up

5.3.1 Broadcast phase

The base station is continuously transmitting the following control and identification information:

- system identify (e.g. country code, operator code, area code etc.);
- system timing information (e.g. slot synchronization, frame synchronization etc.);
- control channel organization and loading information (e.g. announce slot structure especially for random access);
- requests for or denial of system registrations.

Information (such as paging messages addressed to a particular mobile or group of mobiles) is transmitted on a per call basis.

5.3.2 Set-up

Information is exchanged between the infrastructure and mobile. Five elements of the mobile procedure are:

- wake up (if a battery economy mode);

- presence check on control channel (if required);
- transfer to the traffic channel;
- acknowledgement on traffic channel (if required);
- traffic information transfer (voice or data).

Further elements need to be taken into account, especially concerning invoking supplementary services during this phase, conveying this information to the infrastructure, checking the subscriber database to ensure these services have been subscribed to. On successful conclusion of this stage, the mobile progresses to the call in progress stage.

5.3.3 Call in progress

Terminals are now concerned primarily to communicate with each other rather than signal to the infrastructure. However, even during the traffic phase a substantial amount of control information should be supported to allow “traffic channel acknowledgement”, caller authentication, notification of call waiting, call hold and transfer to waiting, priority pre-empt, include call (IC) and speaker identification during a call.

5.3.4 Call clear down

The mobile relinquishes traffic channel and returns to monitoring the control channel. If the call is on “hold” the system will retain details of the mobile and the call reference for subsequent reconnection. The system may optionally retain line resources. When the call is complete all radio and line resources should be cleared of traffic and returned to the resource pool.

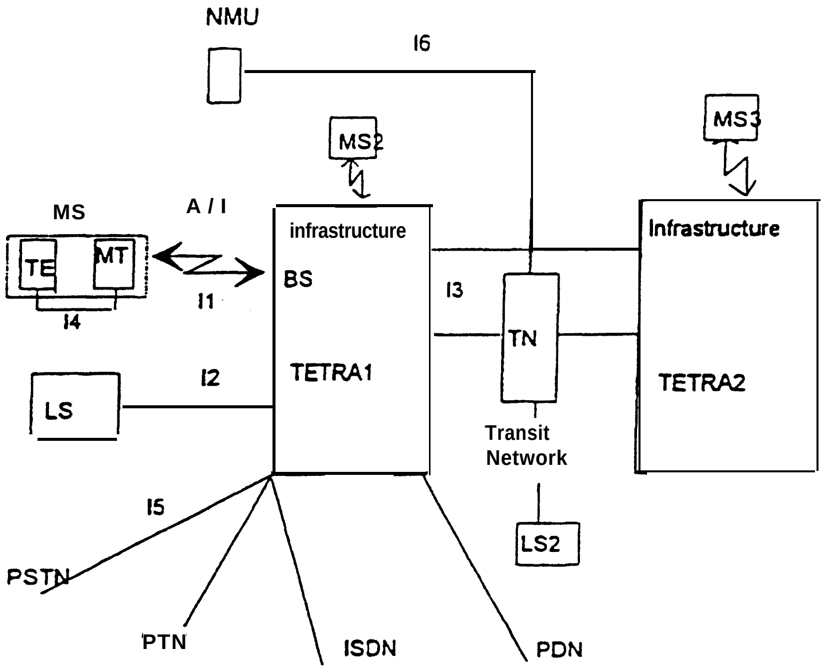
5.4 Connection restoration

A number of network procedures are supported in the TETRA specifications to provide continuity of’ service when a mobile encounters adverse propagation effects, moves between different cells or encounters interference. Connection restoration may also be required for traffic reasons; to redistribute the load on a particular cell such as during minimum mode operation; to allow the frequency allocations at a particular cell to be reorganized, or for maintenance or equipment fault reasons.

The responsibility for initiating the connection restoration procedures can rest with the mobile station or with the base station, depending on the reason for restoration.

The mobile station is responsible for monitoring the quality of the downlink transmissions and may request an alternative channel on the same serving cell if interference is encountered or rnay request service on another cell if the received signal strength drops below a predefined level. The TETRA air interface protocol provides a range of restoration procedures (of different quality) which a network operator may wish to install, and to which users may choose to subscribe. These range from a totally unprepared restoration taking several seconds during which time the connection is broken, to seamless handover where the break in service is imperceptible to the user.

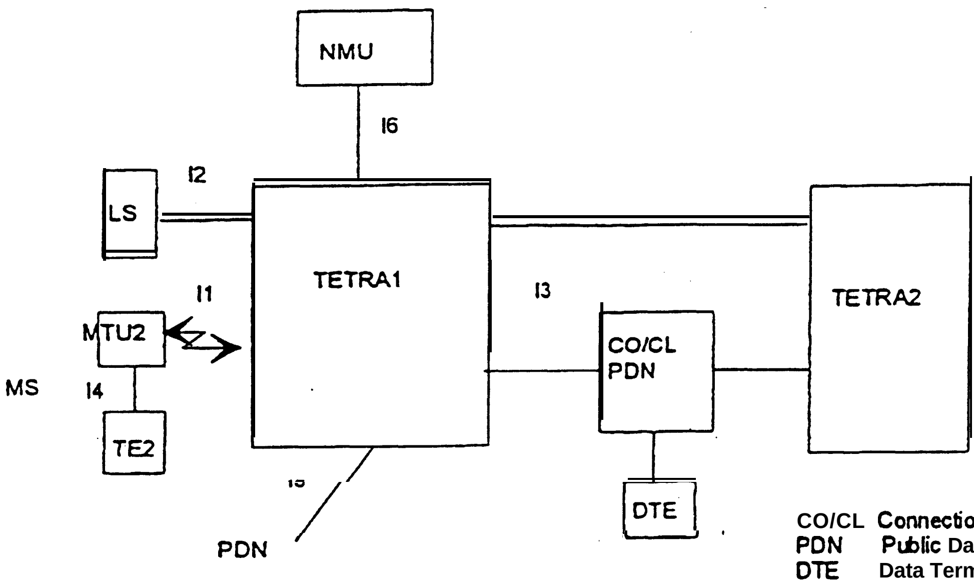
The base station may choose to move the MS to another channel on the same servicing cell if interference on the uplink is encountered. The BS may wish to hand-off the call to an adjacent cell if the loading becomes too high on a particular site (load shedding). This would be performed by altering the acquisition and relinquishing criteria defined in the broadcast (BCCH).



MS Mobile Station
MT Mobile Terminal
LS tine station
NMU Network Management Unit

FIGURE 1

TETRA voice plus data



CO/CL Connection/Connectionless
PDN Public Data Network
DTE Data Terminal Equipment
MTU2 Mobile Terminal Unit
TE Terminal Equipment

FIGURE 2

TETRA packet data optimized

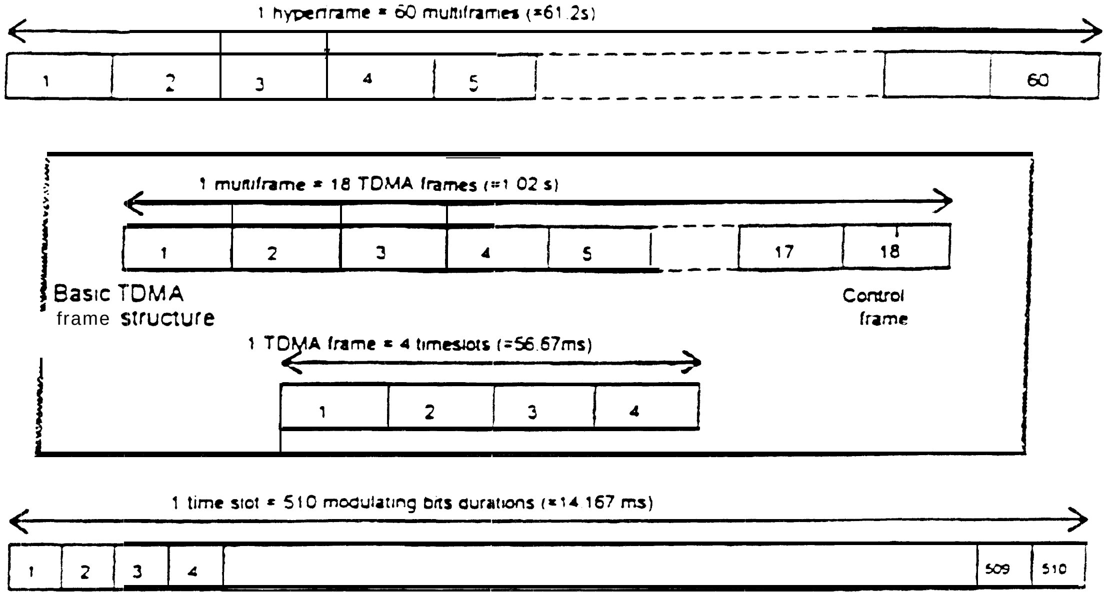


FIGURE 3

TETRA TDMA frame structure

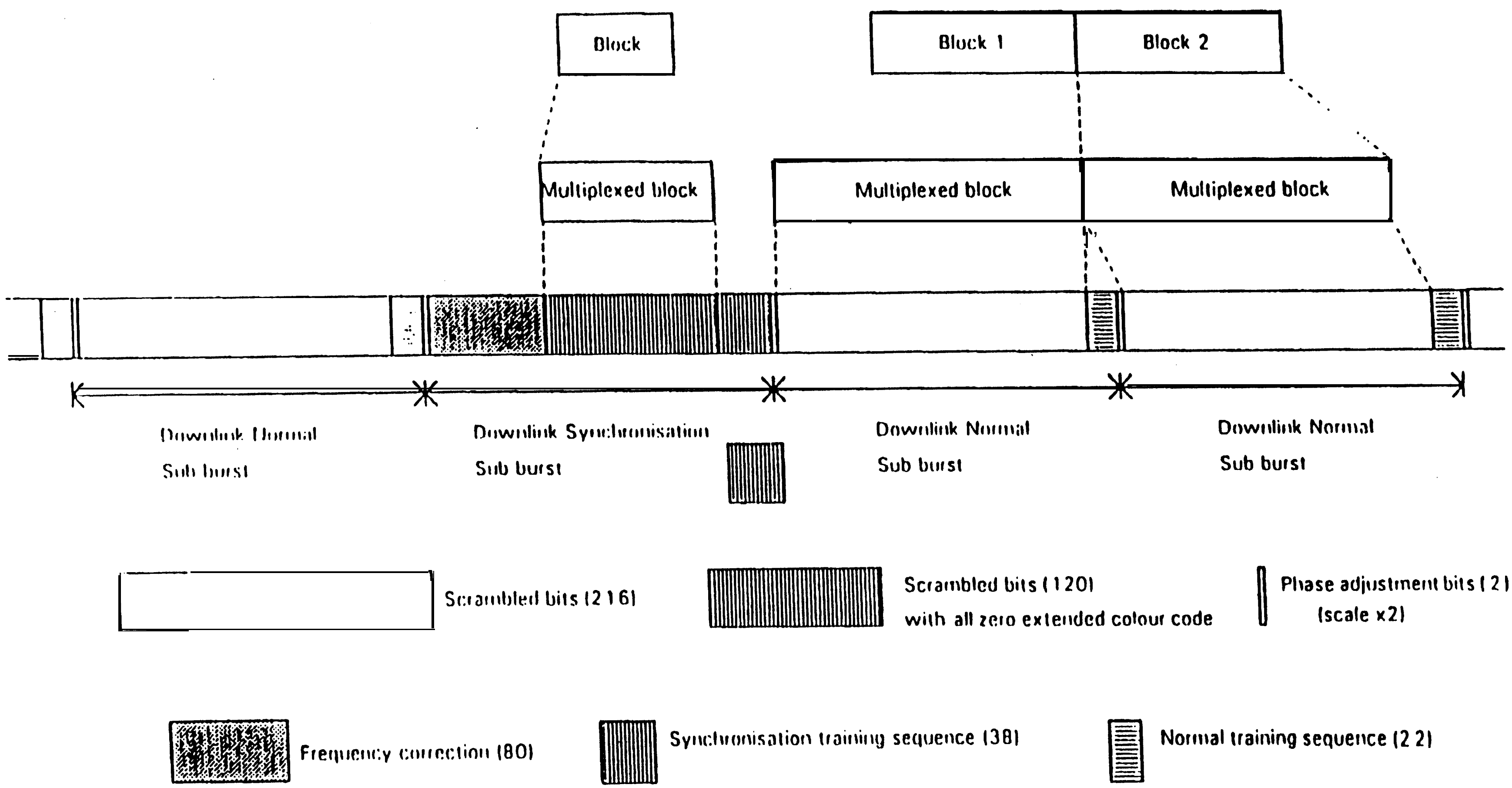


FIGURE 4A

PDO downlink burst structure

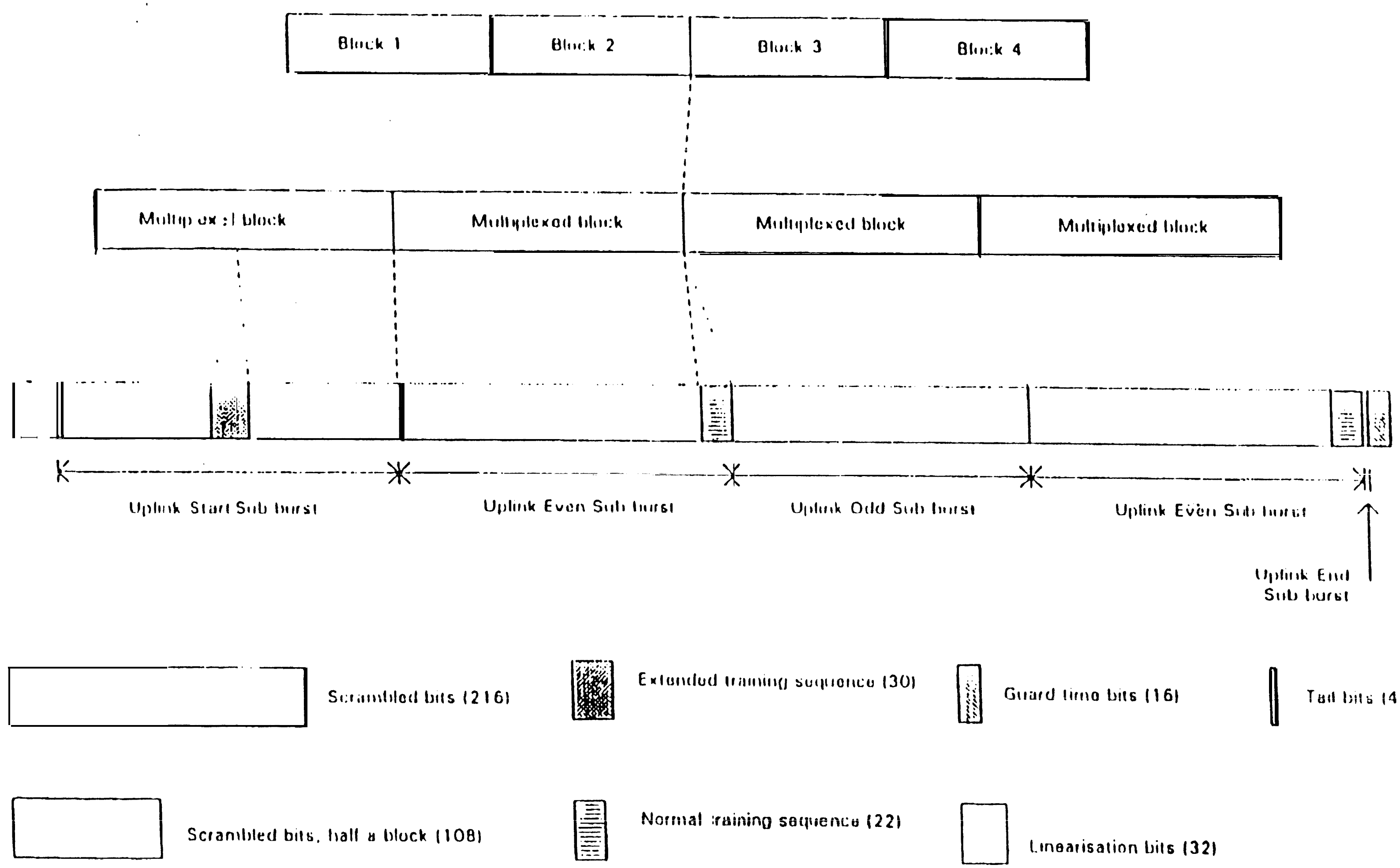


FIGURE 4B

PDO uplink burst structure

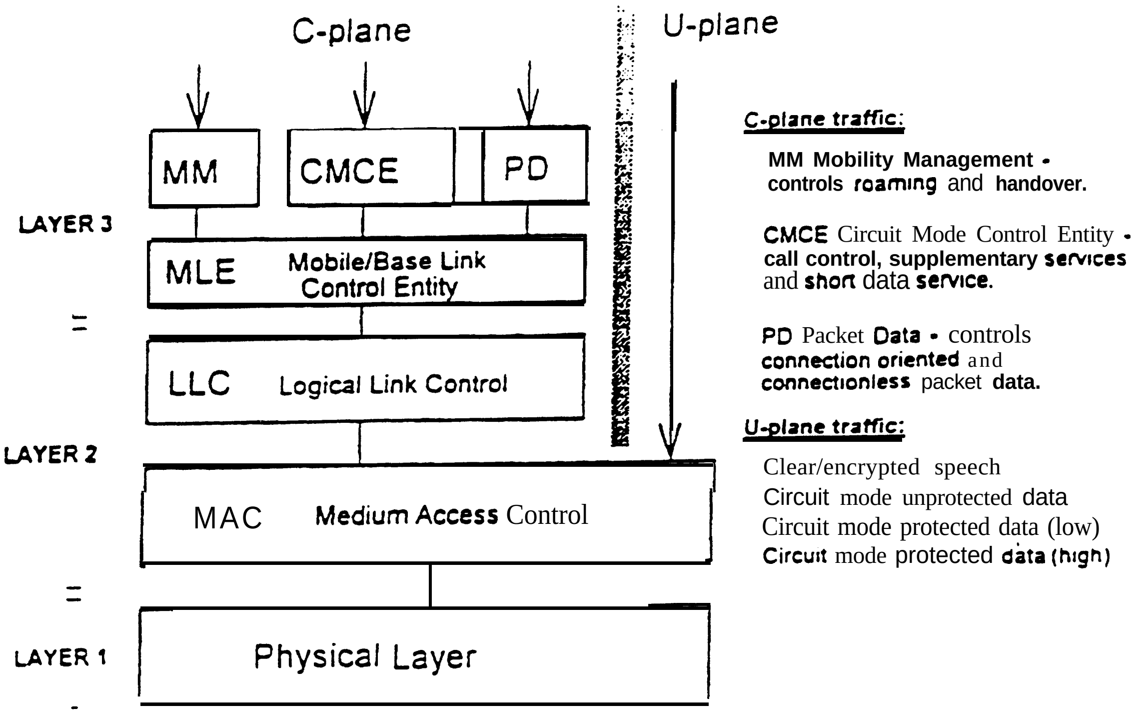


FIGURE 5

Mobile/base station protocol stack

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ETSI [January, 1994] ETR 086-3 – Trans European Trunked Radio (TETRA) system – Technical requirements specification, Part 3 – Security aspects.

ETSI [November, 1994] prETS 300 392-1 – Trans European Trunked Radio (TETRA) – Voice plus Data (V+D), Part 1 – General network design.

ETSI [November, 1994] prETS 300 392-2 – Trans European Trunked Radio (TETRA) – Voice plus Data (V+D), Part 2 – Air Interface.

ETSI [November, 1994] prETS 300 392-10 – Trans European Trunked Radio (TETRA) – Voice plus Data (V+D), Part 10 – Supplementary services stage 1.

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ETSI [November, 1994] prETS 300 393-2 – Trans European Trunked Radio (TETRA) – Packet Data Optimized (PDO), Par-t 2 – Air Interface.

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ANNEX 3

General description of the Project 25 system

1 Services supported

Services will be available on Project 25 systems in accordance with system type and other specifications within this annex. Where a service is mandatory for a Project 25 system type, such a system must provide that service. Where a service is a standard option, and a Project 25 system provides that service, it shall be provided in compliance to the standard. Technological limitations may preclude some systems from supporting certain services.

1.1 Types of systems

Two types of systems are defined: Non-trunked (conventional) and trunked. All Project 25 trunked radios shall be capable of operation in both types of systems.

1.1.1 Non-trunked (conventional)

Non-trunked (conventional) systems possess no centralized management of subscriber operation or capability. All aspects of system operation are under control of the system users. Operating modes within non-trunked systems include both direct (i.e., radio-to-radio) and repeated (i.e., through an RF repeater) operation.

1.1.2 Trunked

Trunked systems provide for management of virtually all aspects of radio system operation, including channel access and call routing. Most aspects of system operation are under automatic control, relieving system users of the need to directly control the operation of system elements.

1.2 Availability

The following tables of telecommunications services show service availability by system type. The services are further denoted as either mandatory or as a standard option, by system type.

Telecommunications services		
Bearer services	**Non-trunked**	**Trunked**
Circuit Switched Unreliable Data	Standard Option	Standard Option
Circuit Switched Reliable Data	Standard Option	Standard Option
Packet Switched Confirmed Delivery Data	Standard Option	Standard Option
Packet Switched Unconfirmed Delivery Data	Standard Option	Standard Option

Teleservices	Non-trunked	Trunked
Broadcast Voice Call	Not Available	Mandatory
Unaddressed Voice Call	Mandatory	Not Available
Group Voice Call	Standard Option	Mandatory
Individual Voice Call	Standard Option	Mandatory
Circuit Switched Data Network Access	Standard Option	Standard Option
Packet Switched Data Network Access	Standard Option	Standard Option
Pre-programmed Data Messaging	Standard Option	Standard Option
Supplementary services	**Non-trunked**	**Trunked**
Encipherment	Standard Option	Standard Option
Priority Call	Not Available	Standard Option
Pre-emptive Priority Call	Not Available	Standard Option
Call Interrupt	Standard Option	Standard Option
Voice Telephone Interconnect	Standard Option	Standard Option
Discreet Listening	Standard Option	Standard Option
Radio Unit Monitoring	Standard Option	Standard Option
Talking Party Identification	Standard Option	Standard Option
Call Alerting	Standard Option	Standard Option
Services to the subscriber	**Non-trunked**	**Trunked**
Intra-system Roaming	Standard Option	Standard Option
Inter-system Roaming	Standard Option	Standard Option
Call Restriction	Not Available	Standard Option
Affiliation	Not Available	Standard Option
Call Routing	Not Available	Standard Option
Encipherment Update	Standard Option	Standard Option

2 Functional groups

2.1 MES (Mobile End System)

In the mobile end system functional group, the term “Mobile” is used as in Land Mobile Radio, which includes all mobile radios, portable radios, and fixed remote radios. The MES functions include the voice and/or data user interface built into a radio.

2.2 MDP (Mobile Data Peripheral)

The mobile data peripheral functional group includes all mobile, portable, and fixed remote data peripherals. The MDP functions include the data user interface of any data peripheral attached to a radio.

2.3 MRC (Mobile Router and Control)

The mobile router and control functional group includes functions of voice and/or data routing, as well as control of the Mobile Radio, MR.

2.4 MR (Mobile Radio)

The mobile radio functional group includes functions of transmission and reception of all RF signals.

2.5 BR (Base Radio)

The base radio functional group includes only the functions of modulation and demodulation of the Radio Frequency energy. Elements within the BR include the Power Amplifier (PA), RF front-end, IF selectivity, and end-IF detection device.

2.6 BA (Base Audio)

The base radio audio functional group includes the functions of frequency/level shaping and signal processing associated with transmitted signals and received signals coupled to the BR. The interface to the BR and BC are manufacturer-specific, and may be at any level or frequency.

2.7 BC (Base Control)

The base radio control functional group includes the automated control functions of an individual radio.

2.8 RFC (Radio Frequency Control)

The radio frequency control functional group includes all logic for translating user command signalling and control into base radio command signalling and control for one or more base radios. The RFC functions further include all logic for generating command signalling and control to a RFS functional group, if present.

2.9 RFS (Radio Frequency Switch)

The radio frequency switch functional group includes all switching for establishing interconnection paths between gateways and base radios, as directed via command and control signalling from an RFC.

2.10 CON (Console)

The CONsole functional group includes all end system functionality for dispatcher(s); including a dispatcher's Man Machine Interface, control and audio functions.

2.11 MSC (Mobile Service Switching Centre)

The mobile service switching centre is a switching centre for services between radio strbnetworks. The MSC is the combination of the RFC and RFS functional groups.

2.12 HLR (Home Location Register)

The home location register is a dynamic database service which tracks the mobility of radios associated with a particular radio subnetwork, that roam to other radio subnetworks.

2.13 VLR (Visitor Location Register)

The visitor location register is a dynamic database service which tracks the mobility of roaming radios which enter a radio subnetwork, but that are associated with a different radio subnetwork.

2.14 RFG (Radio Frequency Gateway)

The radio frequency gateway functional group functions include direct interface with any/all end systems with the exception of the CONsole (where the end system may be an RFG into another radio subsystem), and any translation of command signalling between the end system/user and the RFC. The RFG functions further include any translation of end system/user payload between the user and the RFS. The RFG also includes interface between VLRs, HLRs, and MSCs between RF subsystems.

3 Signalling description

3.1 Data units

Information is transmitted over the air, using the Common Air Interface (CAI), in data units. There are five types of data units defined for voice channel operation, one type of data unit for data packets, and one type of data unit for control functions.

3.1.1 Voice data units

Voice information is transferred in a sequence of Logical Link Data Units (LDUs), each convey 180 ms of voice information. There are two kinds of LDUs, denoted as LDU1 and LDU2. Each LDU conveys additional embedded information, which includes a link control word, an encipherment synchronization word, and low-speed data. LDU1 conveys the link control word. LDU2 conveys the encipherment synchronization word. Both LDU1 and LDU2 convey low-speed data.

Voice information in the LDUs is conveyed as nine frames of vocoder information, with each frame containing 20 ms of digitized voice information.

The LDUs are paired into super-frames of 360 ms. Each superframe has an LDU1 and an LDU2. The last superframe of a voice transmission may terminate after LDU1, if the transmission ends before the LDU2 portion of the superframe has begun. Since LDU2 is present in each superframe (except possibly the last one), it is possible for the transmission recipient to synchronize decipherment in the middle of the transmission, and begin receiving a voice transmission on a superframe boundary.

Voice transmission begins with a header data unit, which conveys the synchronization of the encipherment algorithm. This allows voice information in LDU1 of the first superframe to be deciphered. The header data unit takes 82.5 ms to transmit.

Voice transmission terminates with one of two types of terminator data units. A simple terminator is a short word, 15 ms in duration, signifying the end of a transmission. A terminator with link control conveys a link control word for supervisory functions when terminating a transmission. A terminator with link control is 45 ms in duration.

3.1.2 Packet data unit

A packet data unit conveys general purpose data information. A packet data unit is split into blocks of information. The first block conveys addressing and service information, and is designated as a header block. Subsequent blocks are designated as data blocks. The length of the data packet is contained in the header block.

Each block is protected with either a rate 1/2 trellis code, or a rate 3/4 trellis code. The rate 1/2 trellis code encodes 12 octets of information into exactly 196 bits. The rate 3/4 trellis code encodes 18 octets of information into exactly 196 bits. A header block always uses the rate 1/2 trellis code. Data blocks use a rate 1/2 trellis code for unconfirmed delivery data packets, and a rate 3/4 trellis code for confirmed delivery data packets. The type of data packet (confirmed or unconfirmed) is indicated in the header block.

3.1.3 Control data unit

A special short data packet is defined for control functions. It consists of a single block protected with the rate 1/2 trellis code defined for the packet data unit. It requires 37.5 ms of air time to transmit.

3.2 Media access control

Data units are transmitted over the air preceded by a short burst of frame synchronization and network identity. The frame synchronization is exactly 48 bits, S ms in duration. The network identity is a 64 bit codeword. These allow the recipient of the transmission to determine the beginning of the message, and to distinguish traffic on the proper radio system from interference or co-channel traffic on nearby systems. The network identifier also contains a data unit identifier which identifies among the seven possible data units.

Channel access is controlled with status symbols which are periodically interleaved throughout transmissions. Each status symbol is two bits, transmitted after every 70 bits within a data unit. This spaces the status symbols exactly 7.5 ms apart. The 7.5 ms interval is designated as a microslot time interval. If a data unit happens to end before a microslot boundary, then additional null bits are inserted to pad the transmission to the next microslot boundary.

An RF subsystem indicates activity on an inbound channel by setting the status symbols on the corresponding outbound channel to a “busy” state. Radios wishing to access the inbound channel are inhibited from transmission when the status symbols indicate “busy”. When status symbols indicate “idle”, they may transmit. A third state, indicating “unknown” is used for slotting status symbols.

4 Operational characteristics

Operation over the CAI is dependent on mode, i.e., whether the message is voice or data, and whether the system is trunked or non-trunked. In general, trunked operation requires radios to request service on a control channel using a control data unit. The RF subsystem then assigns the radio to a working channel for further operations. After the operations are complete on the working channel,, the call is cleared for assignment of the channel to other calls. Operation in a non-trunked system does not have the service request phase and the call clearing phase.

4.1 Voice transmit operation

Operation of a transmitter for voice messages has three main cases, with several options and variations of each case. The three main cases consist of Routine Group Calls, Emergency Group Calls, and Individual Calls.

4.1.1 Controls. A transmitter may have several controls which affect transmit operations. Controls sufficient for a radio to support all of the call types are defined below. These controls are:

PTT switch – A Push-To-Talk switch is activated when an operator wishes to transmit, and released when a transmission is finished.

Channel selector – The channel selector is a switch or control that allows the operator of a radio to select a radio's operational parameters. The operational parameters that can be selected include the following items:

1) Transmit frequency.
2) Transmit Network Access Code.
3) Talk Group.
4) Other parameters for setting the vocoder and encipherment functions. For example, the enciphering key variable may be selected.

Emergency switch – The Emergency switch is asserted by a radio operator for emergency calling. Once this switch is asserted, the emergency condition remains asserted until it is cleared by a different means, e.g. turning the radio off.

Numeric keypad/display – This allows a radio operator to set numeric values. This is most useful for individual calls.

4.1.2 Call types. The different types of calls are defined as follows:

Routine group call – This is a transmission that is intended for a group of users in a radio system. Typically, it is the type of call that is made most often. These calls are typically made when the PTT switch is asserted.

Emergency group call – This is a transmission that is intended for a group of users in a radio system, during an emergency condition. The definition of an emergency condition depends on a system's operators, but it typically signifies an exceptional condition with more urgency. These calls are typically made after the emergency switch is asserted.

Individual call -- This is a transmission which is addressed to a specific individual radio. The individual radio's address to which the call is directed is called the destination address. These calls are typically made after the destination address is entered into the radio.

4.1.3 Procedures. The procedures for each of these calls in the transmitter are based on the procedure for the Routine Group Call. Consequently, that type of call is described first, and then the other types of calls are described.

Routine group call procedure

1) PTT. The radio operator asserts the PTT switch.

2) Pre-transmit. The radio selects the channel parameters as determined by the channel selector switch. The radio may check the status symbols, if present, to determine if the channel is busy or idle. If busy, it may optionally hold off the activation of the transmitter until the channel is idle. If the status symbols are not checked, or if the channel is idle, then the radio simply keys the transmitter on the transmit frequency. The radio also activates the voice encoder. The radio also activates the encipherment function, if present.

3) Header Data Unit. The radio transmits the Header Data Unit with the following selected - information fields:

- Network Access Code as determined by the channel selector switch.
- Manufacturer's ID.
- Message Indicator, Algorithm ID, and Key ID are determined by the encipherment function.
- Talk Group/Individual ID is determined by the channel selector switch, as appropriate.

4) Format selection. The following recurrent voice message parameters are set:

- Network Access Code as determined by the channel selector switch.
- Manufacturer's ID.
- Emergency bit is set to indicate routine operation.
- Talk Group/Individual ID is determined by the channel selector switch, as appropriate.
- Source ID is set to the unit ID of the radio.
- Message Indicator, Algorithm ID, and Key ID are determined by the encipherment function.

5) Transmission. The voice link data units, LDU1 and LDU2, are sent with the message parameters set above in step 4. The information contents of the Link Control word is enciphered if specified by the encipherment function. Link Control shall only be enciphered if the voice frames are also enciphered. Transmission is sustained until the PTT switch is released.

6) End of Transmission. Transmission terminates when the PTT switch is released, or some other event forces a dekey, and the transmission has reached the end of an LDU. The radio terminates the voice encoder. Then the radio sends a terminator data unit. A radio always sends the simple terminator, consisting of frame synchronization and the Network ID word. After termination, the radio notifies the encipherment function to terminate, as defined in the encipherment protocol.

7) Dekey. The radio ceases transmission.

Emergency group call procedure

1) Emergency switch. The radio operator asserts the emergency switch. This sets the emergency condition until it is cleared by some other action, e.g., turning the radio off.

2) Group Calls. Activation of the PTT switch now initiates calls that are very much like the Routine Group Call described above. The only difference in procedure is that the Emergency bit is asserted to indicate an emergency condition. Group calls can be made repeatedly, and each group call will indicate the emergency condition.

3) Emergency termination. The emergency condition is cleared by turning the radio off. When the radio is turned on, the emergency condition is cleared and Routine Group Calls are made after PTT assertion. In addition to this method, other methods of termination may also be available.

Individual call procedure

1) Select Called Party. The unit ID of the individual radio to be called can be entered into the radio via a keypad or by some other means. This becomes the destination ID of the call.

2) Make the call. The procedure for group calls is followed, with the following exceptions:

   i) The 'Talk Group ID in the Header Data Unit is cleared to the null talk group (0000).

   ii) The Link Control field is formatted with the individual call format, containing the source ID and destination ID of the call.

4.2 Voice receive operation

The operation of a receiver for voice messages consists of three main cases, with variations that depend on the transmitter's operation. The three main cases are called Squelch conditions in this document. They are: Monitor, Normal Squelch and Selective Squelch.

As in the case of the transmitter, receiver operation will be affected by the channel selector switch. This switch can select:

1) Receive frequency.

2) Receiver Network Access Code.

3) Talk Group.

4) Other parameters for setting the vocoder and encipherment functions. The encipherment function is particularly significant to the receiver.

An additional radio control which can affect a receiver is the monitor switch. This switch allows the operator of a radio to disable any selective squelch of the receiver so that an operator can hear any sign of voice activity. This can be useful for avoiding collisions on non-trunked channels between voice users.

The types of squelch operation described are defined as follows

Monitor – This enables the receiver to unmute on any recognizable voice signal. Selective muting based on the network access code, talk group ID, or unit address is not performed. This is analogous to monitor mode in analogue receivers. This is normally activated with a monitor switch.

Normal squelch – This enables the receiver to unmute on any voice signal which has the correct network access code. Voice messages from co-channel users which are using different network access codes will be muted.

Selective squelch – This mutes all voice traffic except that which is explicitly addressed to the radio. Messages which contain the talk group or unit address of the receiver, as well as the network access code, will be received.

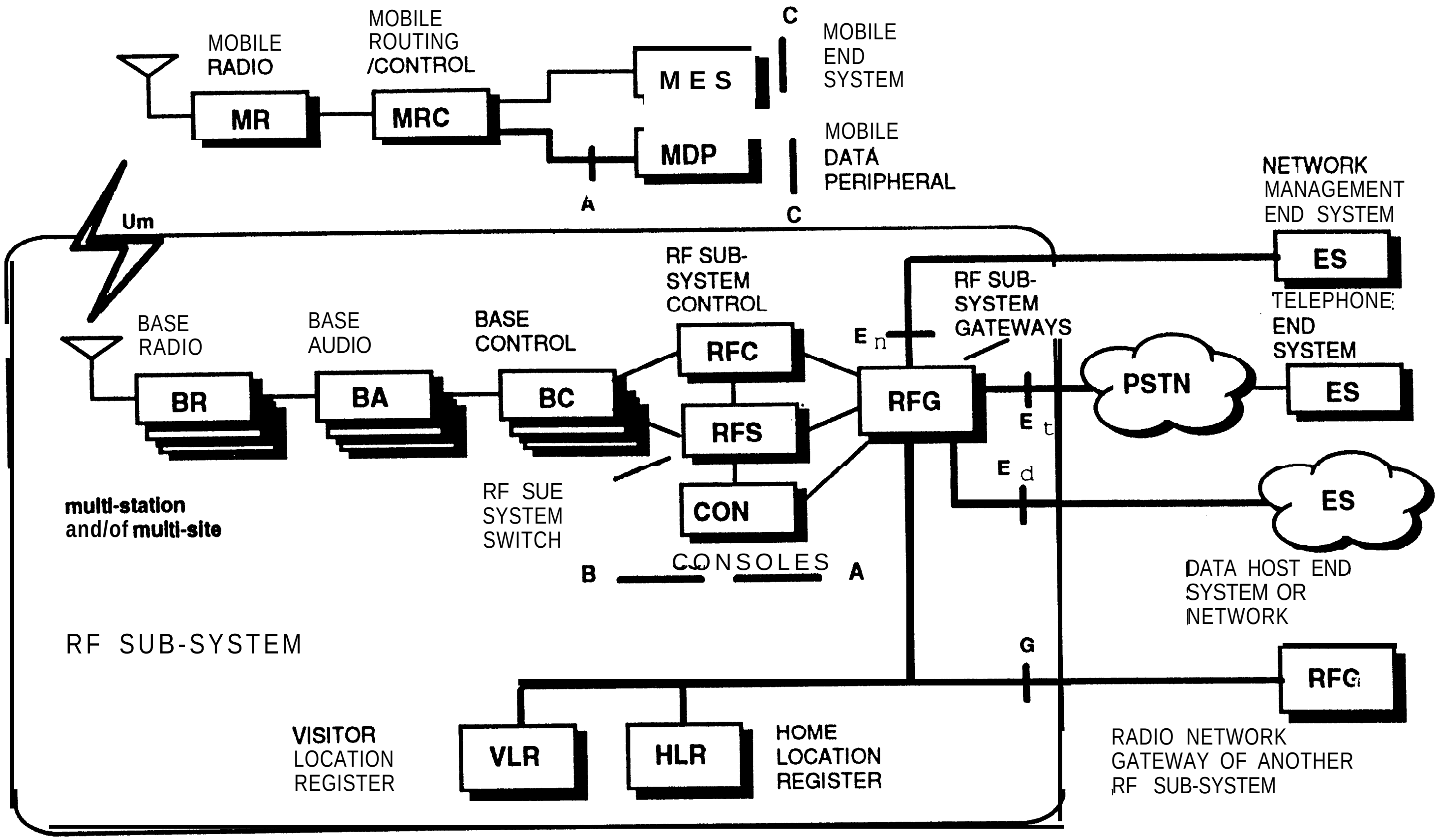

FIGURE 6A

Project 25 Repeater (example) reference configuration

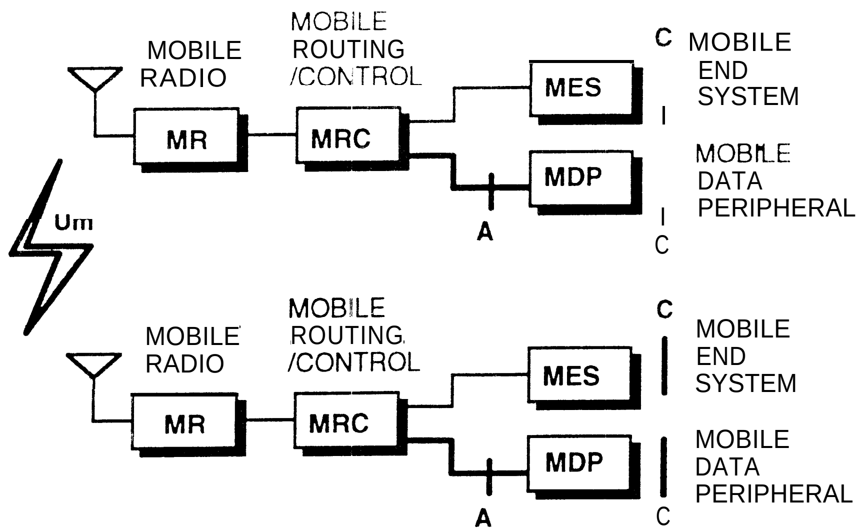

FIGURE 6B

Project 25 Non-repeater reference configuration

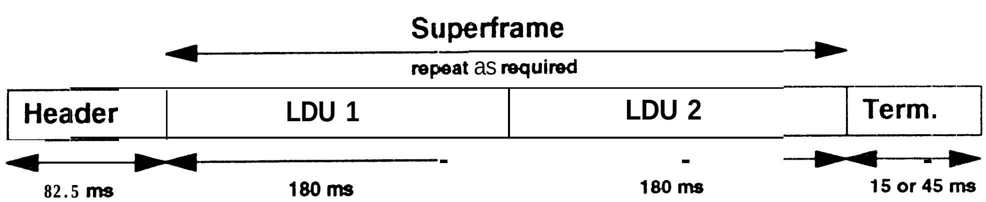

FIGURE 7A

Project 25 voice structure

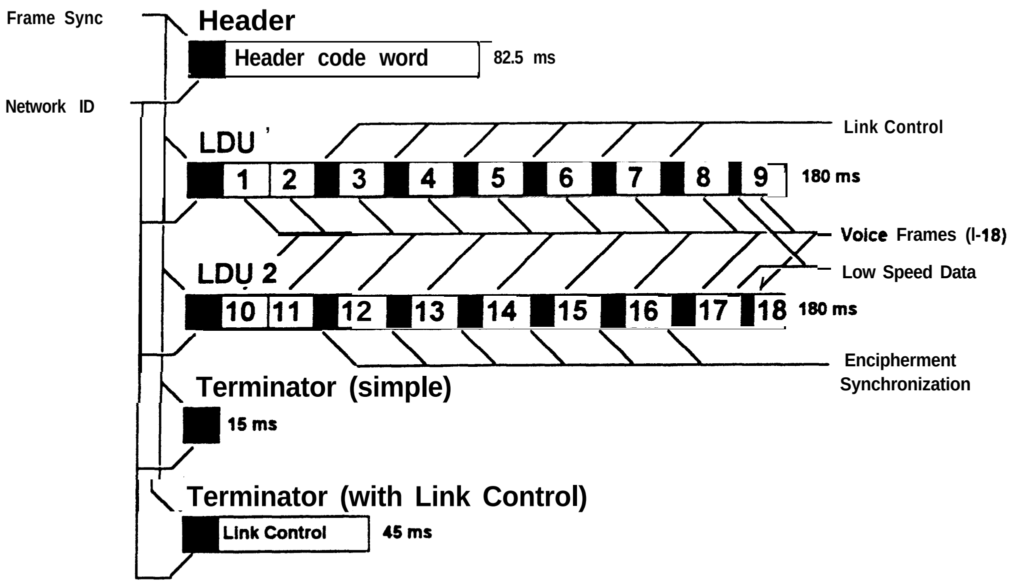

FIGURE 7B

Project 25 voice data unit structure

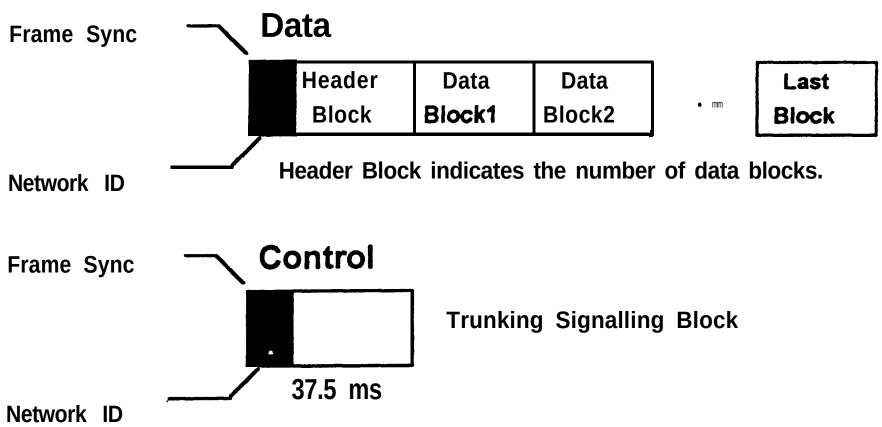

FIGURE 8

Project 25 data and control signal structure

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[2] TWEIA TSB 102.BAAA -- Common Air Interface.

[3] TIA/EIA TSB102.BAAB -- CAI Conformance Testing.

[4] TIA/EIA TSB 102.BAAC -- CAI Reserved Values.

[5] TIA/EIA TSB 102.BAAD – CAI Operational Description for Conventional (non-trunked) Channels.

[6] TIA/EIA IS 102.BABA – Vocoder Description.

[7] TIA/EIA TSB 102.CAAA – Transceiver Measurements and Methods.

[8] TIA/EIA TSB 102.CAAB – Tranceiver Performance Recommendations.

[9] TIA/EIA IS 102.AAAA – DES Encryption Protocol*.

[10] TIA/EIA IS 102.BABB – Vocoder Mean Opinion Score Test.

[11] TIA/EIA IS 102.BABC – Vocoder Reference Test.

[12] TIA/EIA TSB 102.BABD – Vocoder Selection Process.

[13] TIA/EIA TSB102.AABA – Trunking Overview.

[14] Draft TIA/EIA TSB102.AABB – Trunking Control Channel Formats.

[15] Draft TIA/EIA TSB102.AABC – Trunking Control Channel Messages.

[16] TIA/EIA TSB 102.BAEA – Data Overview.

[17] TIA/EIA TSB 102.BAEB – Packet Data Specification.

[18] TIA/EIA TSB 102.BAEC – Circuit Data Specification.

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[24] Draft TIA/EIA IS 102.BADA – Telephone Interconnect Requirements and definitions (voice service).

[25] Draft TIA/EIA TSB102.AABF – Link Control Words.

[26] Draft TIA/EIA TSB 102.AABG – Conventional Control Messages.

[27] Draft TIA/EIA TSB 102.AABD – Trunking Procedures.

[28] TIA/EIA TSB 102.AACB – OTAR Operational Description*.

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* These documents are referenced for completeness only. The selection of encipherment algorithm should remain a national option.

ANNEX4

General description of the IDRA System

1 Introduction

The IDRA System has been developed for use mainly in business-oriented mobile communications applications. Both voice and data communications in the IDRA System offer inter-mobile communications in a single cell and inter-mobile communications between cells, as well as communications between a PSTN user and a mobile subscriber to the IDRA. The IDRA System satisfies the following three fundamental specifications:

1) Voice only.

2) Voice and data (Circuit mode data, short message mode data, and packet mode data).

3) Data only (Circuit mode data, short message mode data, and packet mode data).

2 Services

2.1 Teleservices

Clear speech or enciphered speech in each of the following:

- Individual call (point-to-point).
- Group call (point-to-multipoint).
- Broadcast call (point-to-multipoint, one way).

Full-duplex interconnect call.

Full-duplex dispatch call (option).

2.2 Bearer services

Individual call, group call, and broadcast call for each of the following:

- Circuit mode protected data 3.044 and 4.8 kbit/slot.
- Circuit mode non-protected data 7.466 kbit/slot.
- Packet connectionless data.
- Packet connection-oriented (option).

2.3 Supplementary services

Telephone type supplementary services:

- Call completion to busy/no-reply subscriber.
- Call barring incoming/outgoing call.
- Calling line identity presentation.
- Calling line identity restriction.
- Voice operation guide (option).
- List search call (option).

- Call waiting
- Advice of charge (option)
- Short message service (option).
- Call traffic monitor.
- Call monitor with late entry
- Priority call.
- Conference call (option).
- Area selection.
- Subgrouping Call.

Network access supplementary services:

- Multiple-zone access.
- PSTN/PSDN access.

2.4 Security aspects

Special security aspects are not specified, but the system provides a high level of security with authentification and identification.

Authentification

During power up, mobile origination, mobile termination, location updating, supplementary service, and/or short message service.

Identification

By individual identification and/or temporary identification.

3 Overview of the system

The network approach showing the major architectural components of the system is shown in Figure 9.

4 System specifications

Refer to Table 1.

4.1 Logical channels

The following logical channels are defined:

- Broadcast Control Channel (BCCH).
- Common Control Channel (CCCH).
- Associated Control Channel (ACCH)
- Traffic Channel (TCH).
- Packet Channel (PCH).
- Slot Information Channel (SICH).

- Random Access Channel (RACH).
- Temporary Control Channel (TCCH).
- Dedicated Control Channel (DCCH).
- Radio Control Channel (RCCH).

4.2 TDMA frame structure

The basic frame is prescribed at six slots. The corresponding outbound and inbound frames make a pair. The frame offset, the outbound frame delay relative to the inbound frame, is 70.955 ms.

Conversely, the inbound frame delay, relative to the outbound frame (referred to as transmit-receive offset) can be calculated by the formula, (frame length)-(frame offset). Accordingly, transmit-receive offset is 19.045 ms. Figure 10 shows the general frame structure of the IDRA System.

4.3 Traffic channels

4.3.1 Speech traffic channels

The speech codec for voice communication services, including error correction and error detection mechanisms, has not been defined in the ARIB standard. However, the ARIB defined the frame structure of the voice channel to have 90 ms speech frames comprised of a total of 672 bits, including the additional bits for error correction. The system operator is free to choose the codec bit rate and error control scheme up to a total of 7.467 kbit/s.

4.3.2 Data traffic channels

A circuit data protocol is available for circuit data applications. The circuit-switched data protocol offers a full-duplex packet stream.

Packet data transmission is a planned feature of the IDRA. Airtime for packet transmission is dynamically allocated to the user devices according to their instantaneous communication need. The packet data protocol is planned to allow an auto-bauding capability so that different net burst transfer rates will be available to the user.

5 Operational characteristics

5.1 Location updating and roaming

5.1.1 Roaming

Roaming, which enables automatic switching of the infrastructure when a Mobile Station (MS) moves into a different location area, is possible between IDRA Systems.

5.1.2 Location updating (option)

The IDRA System tracks an individual MS's location to allow the MS to move freely throughout the system and receive or originate calls. Location areas, which are composed of one or more sites, are used to define geographical areas in the system. The mobile terminal must report its position each time it moves between location areas.

5.1.3 Handover (option)

The IDRA supports handover between zones and between systems. Handover allows for maintaining the link quality for user connections, minimizing interference, and managing traffic distributions

5.2 Communication protocols

The communication protocols of the IDRA are layered according to the OSI model as shown in Figure 11. However, it does not strictly match the standard model because press-to-talk communication is the basic operation, so a protocol providing a faster response is required.

The layers are subdivided as shown below:

- Layer 1: This layer specifies the physical structure of the channel. (Basic slot format, subslot format, etc.).
- Layer 2: This layer specifies communication control between the MS and the infrastructure such as random access control, polling control and time alignment control.
- Layer 3: This layer performs as a network layer and is divided into the following three sublayers:
  - Connection Management (CM)

    Call set-up, Call management/control, Call clear down, etc.
  - Mobility Management (MM, option)

    Location Registration, Authentication, etc.
  - Radio Resource Management (RR, option)

    Cell selection, channel assignment, handover, etc.

5.3 Call set-up

5.3.1 Broadcast phase

The base station is continuously transmitting the following control and identification information:

- Control channel information (e.g. physical structures of control channel for system identification and call set-up).
- System information (e.g. types of communication services and protocols which IDRA can provide).
- Restriction information (e.g. types of communication services and protocols which IDRA now restricts).
- System structure information (e.g. location area and target cell information; optional).

5.3.2 Set-up

Necessary information is exchanged between the infrastructure and MS. The elements of the mobile procedures are:

- Wake up (if in battery saving mode).
- Receive the control channel.
- Exchange the necessary information for call set-up.
- Receive the traffic channel.

Transfer traffic information (voice or data).

- Registration and authentication (option).

5.3.3 Call clear down

The following six procedures are available for call clear down:

The MS and the infrastructure clear down when the time limit for communication is reached.

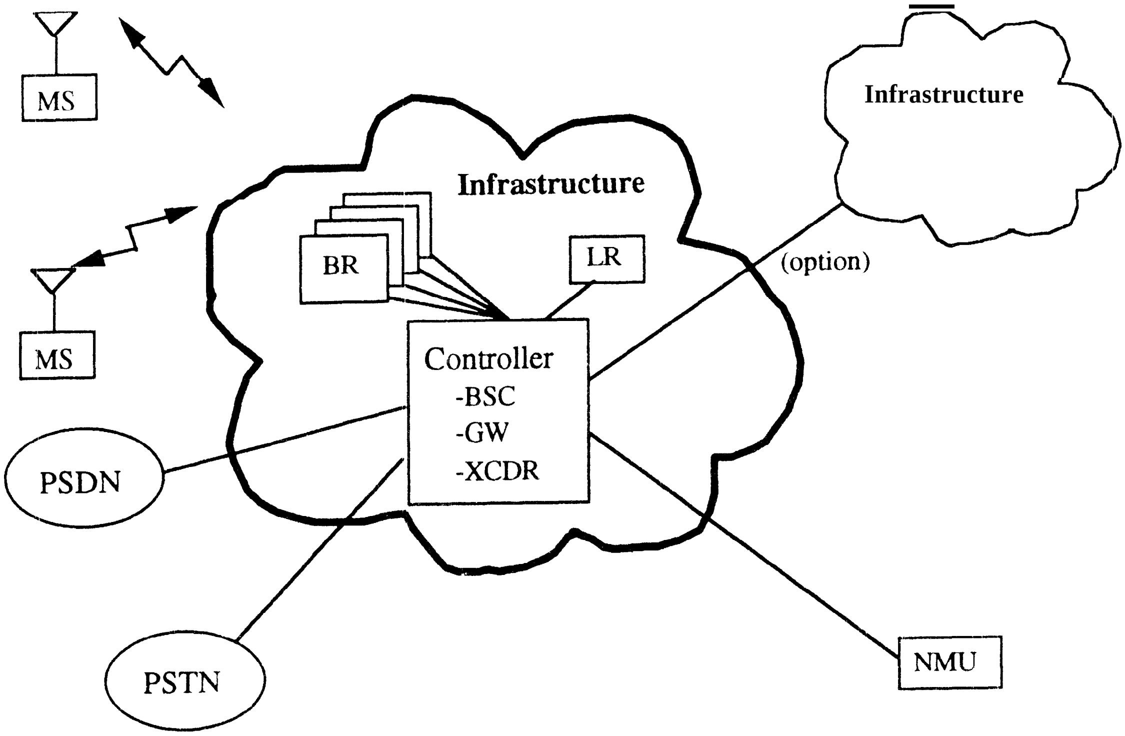
The infrastructure clears down when the time limit for no response is reached.

The infrastructure clears down when the time limit for no communication is reached.

- The MS clears down on detection of poor traffic conditions.
- Clear down occurs on demand of disconnection from a mobile terminal, a fixed terminal, or a telephone on the PSTN.
- Disconnection from the base.

5.4 Connection restoration (option)

- MS knows where to monitor from information on Broadcast Control Channel.
- MS continuously measures parameters during call:
  - C/(I + N).
  - RSSI.
  - Primary serving channel.
- When MS detects trouble on primary server:
  - MS sends in parameter samples.
  - Base evaluates potential servers.
  - Base assigns new server.
  - MS switches to new server.



BR	: Base Radio
BSC	: Base Site Controller
LR	: Location Register
GW	: Gate Way
X:CDR	: Speech Transcoder
NMU	: Network Management Unit
MS	: Mobile Station
PSTN	: Public Switched Telephone Network
PSDN	: Public Switched Data Network

FIGURE 9

IDRA Network approach

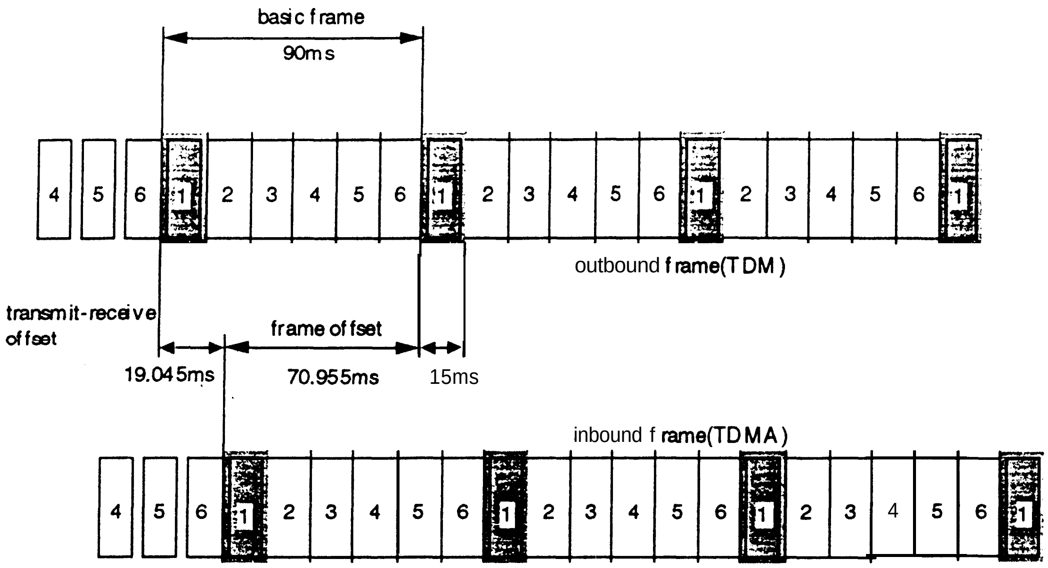


FIGURE 10

IDRA TDMA Frame structure

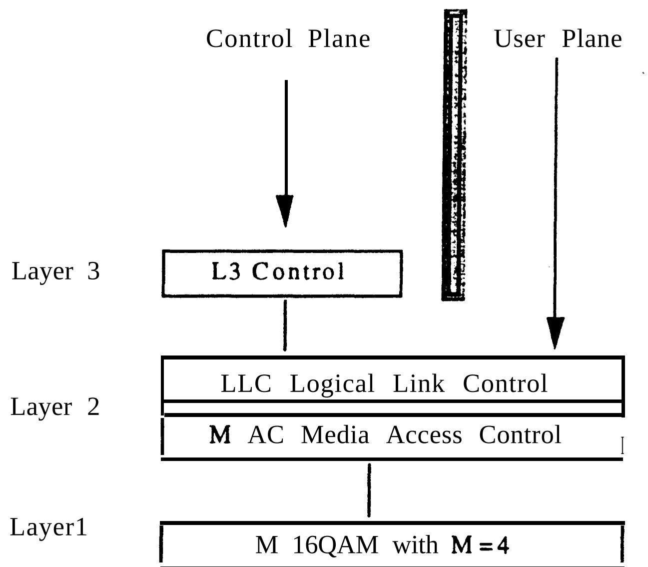


FIGURE 11

Protocol stack

REFERENCE

ARIB (November, 1995) RCR STD-32A – Integrated Dispatch Radio System.

ANNEX 5

General description of the DIMRS System

1 Introduction

The Digital Integrated Mobile Radio System (DIMRS), using new digital technology, fully integrates multiple services including, radio-telephone, paging and dispatch communications into a single infrastructure. DIMRS caters both to users who require an integrated system with enhanced services as well as users who cannot justify the use of a separate pager, cellular phone, dispatch radio and data modem.

2 System services

The services provided are:

Dispatch

1) Group call.
2) Private call.
3) Call alert.
4) Push-to-Talk (PTT) ID.
5) Landline to individual private call.
6) Selective "area" calling.

Interconnect

1) Interconnect with other switched networks.
2) Full-duplex operation.
3) Handover.
4) Custom calling features (call waiting, three party calling, DTMF access to services, call forwarding, busy transfer, no answer transfer, call restrictions, access to information services).

Roaming services

1) Intra-system roaming.
2) Inter-system roaming.
3) System-to-system handover.
4) Inter-system calling features.
5) Registration/de-registration.

Message paging

1) Paging.

2) Short message service.

Data communications

1) Circuit mode (protected).

2) Packet mode:
   - with handshake;
   - without handshake.

3 Authentication mechanism

DIMRS provides system security control with an authentication mechanism which may be invoked prior to any chargeable service initiation.

Authentication is used to verify that a mobile station is registered in the system. It may take place during the location updating, mobile origination, mobile termination, supplementary service, and short message service procedures for an interconnect subscriber. For a dispatch only subscriber, authentication will occur during power-up or when a subscriber crosses certain system boundaries such as into another service provider's area.

Each Mobile Station (MS) user is assigned an individual ID, referred to as an International Mobile Subscriber Identity (IMSI), which is understood by both the dispatch and interconnect call processing programmes. The system will validate the user IMSI each time an interconnect call processing procedure is performed.

For interconnect call processing, a temporary ID, referred to as the Temporary Mobile Station Identifier (TMSI), is used to identify the MS to the system. This minimizes broadcasting the IMSI over the air.

4 Overview of the system

The network approach showing the major architectural components of the system is shown in Figure 12.

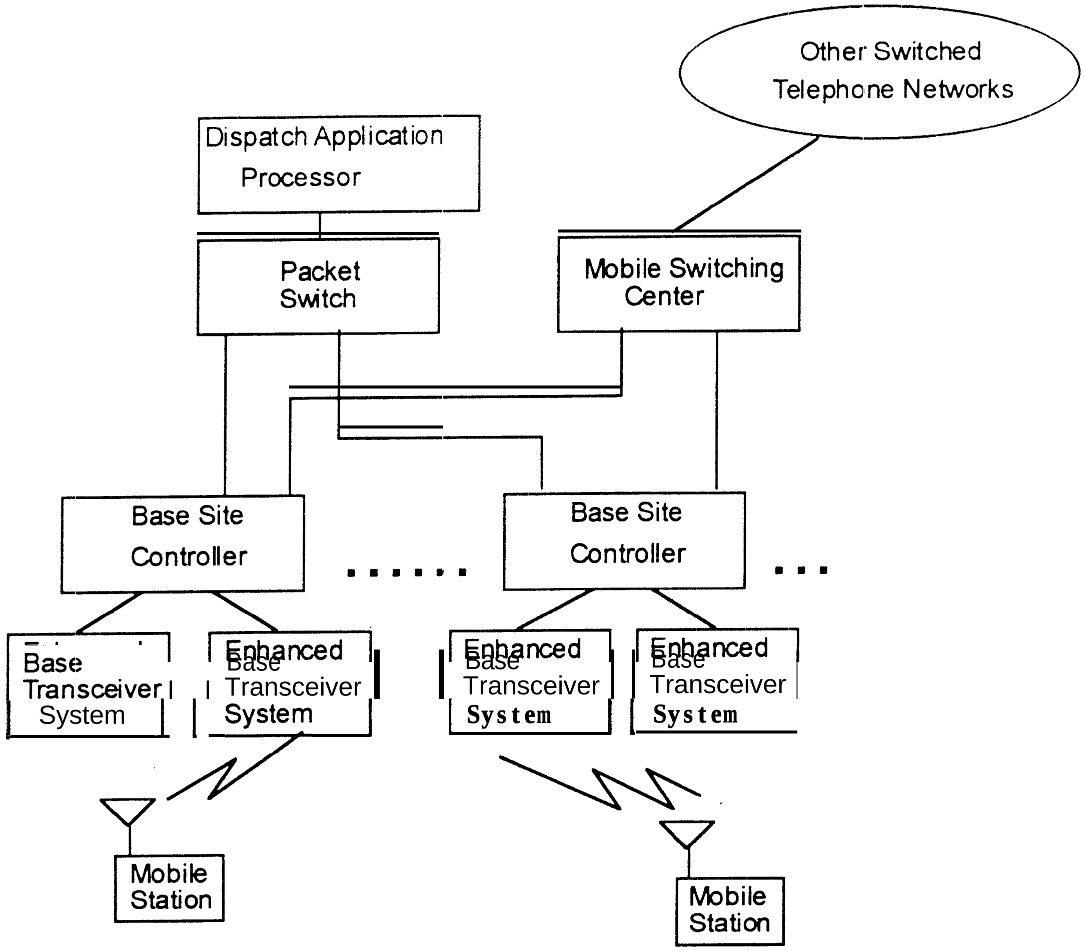


FIGURE 12

DIMRS Network approach

5 System specifications

Refer to Table 1.

5.1 Logical channels

The following logical channels are defined:

Slot Information Channel (SICH)

A broadcast channel used for transmission of slot control information.

Primary Control Channel (PCCH) comprising:

- Broadcast Control Channel (BCCH).
- Common Control Channel (CCCH).
- Random Access Channel (RACH).

The PCCH is a multiple access channel used for Layer 3 control signalling between the fixed network equipment and the mobile stations. Each cell has one PCCH.

Temporary control channel (TCCH)

A temporarily allocated multiple access channel used to provide a means for inbound random access on a channel which is normally reserved access.

Dedicated control channel

Supports more extended Layer 3 control procedures which would be inefficient if conducted on the PCCH.

Associated control channel (ACCH)

The ACCH provides a signalling path on the traffic channel. The main application of the ACCH is to support whatever Layer 3 control signalling is required for traffic channel supervision. Bandwidth for the ACCH is obtained by dynamically stealing on the TCH.

Traffic channel (TCH)

- Circuit-Switched Channels

  These channels are used to transport voice or circuit-switched data traffic.
- Packet-Switched Channel (PCH)

  These channels will support packet-switched user data communications.

5.2 TDMA frame structure

The DIMRS data stream structure, shown in Figure 13, has six slots per TDMA cycle. A frame structure is further superimposed on this cyclical structure. Inbound and outbound frames consist of 30 240 slots, each 15 ms long. The duration of the frame is 453.6 seconds.

A hyperframe structure is also defined, in addition to the frame structure. A hyperframe comprises 256 frames, thus, it contains a total of 7 741 440 slots and has a duration of 116 121.6 seconds (32 hours, 15 minutes, 21.6 seconds). The large number of slots in the hyperframe is useful for implementing encryption.

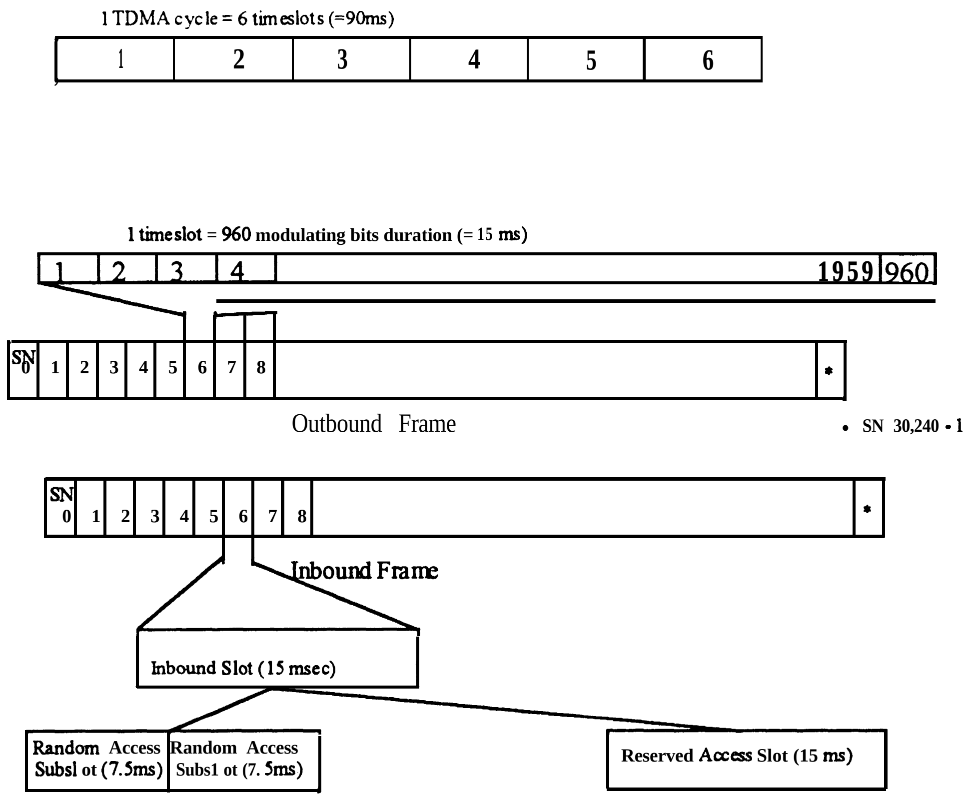


FIGURE 13

DIMRS Frame structure

5.3 Traffic channels

5.3.1 Speech traffic channels

The speech coding technology used is Vector Sum Excited Linear Prediction (VSELP). Acceptable quality is maintained at channel bit error ratios as high as 4-5% in Rayleigh fading, or 10% in static conditions. Error correction is realized through a variable rate strategy whereby the uncoded and trellis-coded 16 QAM modulations are applied selectively to speech bits in accordance with their perceptual significance.

5.3.2 Data traffic channels

A circuit data protocol is available for circuit data applications such as laptop or palmtop computers, fax and image processing, and file transfer applications. The circuit-switched data protocol offers a full-duplex packet stream with a single rate of 7.2 kbit/s (six users per RF carrier). This includes forward error correction coding and selective re-transmission of non-correctable blocks.

Allowance has been made for packet data in DIMRS. Bandwidth will be dynamically adjusted to accommodate demand.

6 Operational characteristics

6.1 Location updating and roaming

6.1.1 Intra-system roaming

DIMRS tracks a unit's location so that calls can be routed to it. Both the dispatch and interconnect calls require the current location of an MS. The DIMRS system will utilize a Location Area (LA). The unique identity of a location area is conveyed via cyclic broadcast on the primary control channel. The mobile monitors the preferred primary control channel and issues a location update request when it finds its location area is no longer supported. The location update request is sent to the Visitor Location Register (VLR) that holds the current location of MS units operating in that system.

6.1.2 Inter-system roaming

The ability to travel freely throughout the single service area and originate or receive calls without regard to current location can be extended to allow MS's to travel from one service area to another. A single service area can consist of multiple cells covering a large geographical area (e.g. entire metropolitan area). Alternatively, it may be-necessary or desirable to subdivide it into multiple service areas, because of RF coverage gaps, management, or regulatory issues.

6.1.3 System-to-system handover

DIMRS supports handover between cells, between Location Areas, and between systems. Handover allows for maintaining the link quality for user connections, minimizing interference, and managing traffic distributions. The inter-system handover is facilitated in the MS's switch.

6.1.4 Inter-system calling features

The MS's in the DIMRS can achieve inter-operability between any system configurations.

6.2 Communication protocols

The communication protocols are layered according to the Open Systems Interconnection (OSI) reference model.

6.3 Operation

6.3.1 Dispatch call operation

1) A dispatch call is requested via PTT activation.

The call request packet is routed to the Dispatch Application Processor (DAP).

The DAP recognizes the MS unit's group affiliation and tracks the group members' current location area.

2) The DAP sends location requests to each group member's location area to obtain current sector/cell location.

3) The MS units in the group respond with current sector/cell location.

4) The DAP instructs the originating EBTS with packet routing information for all group members.

5) Call voice packets are received by the Packet Duplicator (PD), replicated, and distributed to the group's end nodes.

6.3.2 Telephone interconnect operation

A Call initiation – Inbound

1) Random Access Procedure (RAP) on primary control channel.

2) Get dedicated control channel assigned.

3) Authentication (optional).

4) Call setup transaction.

5) Get assigned to a traffic channel.

6) Talk.

7) Call termination request on associated control channel.

8) Channel released.

B Call initiation – Outbound

1) Page MS on primary control channel.