NATO STANDARDIZATION OFFICE
BUREAU OTAN DE NORMALISATION

18 January 2018 NSO/0066(2018)CAP1/4622

STANAG 4622 CAP1 (EDITION 1) - INTEROPERABILITY STANDARD FOR SATELLITE BROADCAST SERVICES (SBS)

References:
A. AC/322(SC/6)N(2008)0031, dated 29 March 2008 (Edition 1) (Ratification Draft 1)
B. AC/322-N(2017)0168-AS1 dated 10 January 2018

1. The enclosed NATO Standardization Agreement, which has been ratified by nations as reflected in the NATO Standardization Document Database (NSDD), is promulgated herewith.

2. Reference A. is to be destroyed in accordance with local document destruction procedures.

ACTION BY NATIONAL STAFFS

3. National staffs are requested to examine their ratification status of the STANAG and, if they have not already done so, advise the NSO, through their national delegation as appropriate of their intention regarding its ratification and implementation.

4. It should be noted that this standard entered development/ratification under AAP-03(I) and therefore is promulgated in its current format as approved by the related Tasking Authority in accordance with Reference B.

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Enclosure:
STANAG 4622 (Edition 1)

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NORTH ATLANTIC TREATY ORGANIZATION
(NATO)

NATO STANDARDIZATION OFFICE
(NSO)

STANDARDIZATION AGREEMENT
(STANAG)

SUBJECT: INTEROPERABILITY STANDARD FOR SATELLITE BROADCAST SERVICES (SBS)

Promulgated on 18 January 2018

Edvardas MAŽEIKIS
Major General, LTUAF
Director, NATO Standardization Office

RECORD OF AMENDMENTS

No.	Reference/Date of amendment	Date entered	Signature

EXPLANATORY NOTES

AGREEMENT

1. This STANAG is promulgated by the Director NATO Standardization Office under the authority vested in him by the North Atlantic Council.

2. No departure may be made from the agreement without informing the tasking authority in the form of a reservation. Nations may propose changes at any time to the tasking authority where they will be processed in the same manner as the original agreement.

3. Ratifying nations have agreed that national orders, manuals and instructions implementing this STANAG will include a reference to the STANAG number for purposes of identification.

RATIFICATION, IMPLEMENTATION AND RESERVATIONS

4. Ratification, implementation and reservation details are available on request or through the NSO websites (internet http://nso.nato.int; NATO Secure WAN http://nso.hq.nato.int).

RESTRICTION TO REPRODUCTION

FEEDBACK

6. Any comments concerning this publication should be directed to NATO/NSO – Bvd Leopold III - 1110 Brussels - Belgium.

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C – Quality of Service (QoS)
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F – Analog Video and Audio Encapsulation
G – Data Encryption
H – DVB Modulation
I – Return Channel for Interactive Services
J – Second Generation DVB Modulation

Related Documents:

ETSI TR 101 154 V1.4.1 (2000-07) Technical Report Digital Video Broadcasting (DVB) Implementation guidelines for the use of MPEG-2 Systems, Video and Audio in satellite cable and terrestrial broadcasting applications

ETSI TR 102 154 V1.1.1 (2001-04) Technical Report Digital Video Broadcasting (DVB) Implementation guidelines for the use of MPEG-2 Systems, Video and Audio in Contribution and Primary Distribution Applications

EN 300 421: DVB: Framing Structure, Channel Coding, and Modulation for 11/12 GHZ Satellite Services

EN 301 192: Digital Video Broadcasting: DVB Specification for data broadcasting

EN 301 790: Digital Video Broadcasting (DVB): Interaction channel for satellite distribution systems

EN 302 307: Digital Video Broadcasting (DVB): Second generation framing structure, channel coding, and modulation systems for Broadcasting, Interactive Services, News Gathering, and other broadband satellite applications

ISO/EIC 13818-1: Generic Coding of Moving Pictures and associated audio information: Systems

ISO/EIC 13818-6: IT – Generic Coding of moving pictures and associated audio information: Extensions to DSM-CC

RFC 4303: IP Encapsulating Security Payload

RFC 4305: Cryptographic Algorithm Implementation Requirements for Encapsulating Security Payload and Authentication Header

RFC 1301: Multicast Transport Protocol

RFC 768: User Datagram Protocol

RFC 793: Transmission Control Protocol

RFC 1771: A Border Gateway Protocol 4

RFC 2784: Generic Routing Encapsulation

RFC 3077: A Link Layer Tunneling Mechanism

SatLabs Group: SatLabs System Recommendations

1. (NU) AIM

The Satellite Broadcast Service (SBS) will provide NATO forces with high capacity satellite communications for strategic and tactical units. The aim of this agreement is to define a baseline set of interfaces and protocols so that system integrators are able to build and commission interoperable SBS transmit and receive platforms using COTS based satellite broadcast technology and NATO approved HAIPIS-compliant Type-1 IP-encryption devices.

2. (NU) AGREEMENT

The requirements specified in this STANAG are meant to ensure a basic commonality amongst NATO broadcast satellite systems. Participating nations agree to use the characteristics contained in this STANAG for development and deployment of Satellite Broadcast Service (SBS) systems.

3. (NU) GENERAL

The details of this agreement are defined in Section 5.0 and Annexes A-J. Annex A provides a list of Terms, acronyms and abbreviations of significant meaning in the context of this STANAG. Annex B identifies standard IP protocols needed to support unicast/multicast routing and IP tunneling. Appendix C identifies Quality of Service (QoS) implementation. Annex D provides a description of standards based formats for IP over MPEG-2 encapsulation. Annex E provides a description of standards based formats for serial stream data transfer over MPEG-2 encapsulation. Annex F provides a description of standards based formats for Analog Video and Audio data over MPEG-2 encapsulation. Annex G provides a description of Data Encryption mechanisms. Annex H provides a description of the Digital Video Broadcast (DVB) waveform and modulation parameters. Annex I provides a description for implementation of a return channel for interactive services. Annex J provides a description of the second generation DVB waveform and modulation parameters.

4. (NU) DETAILS OF THE AGREEMENT

NATO SBS systems shall consist of a transmit gateway equipped with the appropriate interfaces necessary to communicate with external networks for which it will obtain its broadcast information. The transmit gateway shall encapsulate IP, Analog A/V (if applicable), and Serial Stream (if applicable) data into an MPEG-2 Transport Stream for modulation into the Digital Video Broadcast (DVB) waveform. Annexes A-H provide a description of the standards required for each respective data type to be prepared for over-the-air transmission via DVB. The transmit gateway shall have the capability to allocate bandwidth based on the type of data being broadcast, ensuring that no single type or class of data saturates the link. The transmission gateway shall have the ability to encrypt data at designated security levels using NATO approved type-1 cryptographic devices.

NATO SBS systems shall consist of a receive gateway equipped with the appropriate interfaces necessary to communicate with external networks for which it will provide broadcasted information. The receive gateway shall have the ability to parse data received from the transmission gateway by applying the inverse order of standards used to format the broadcasted data (Annexes A-J). The receive gateway shall have the ability to decrypt data at designated security levels using NATO approved type-1 cryptographic devices.

Internal interfaces and specific components of the transmission and receive gateways are not specified in this STANAG and will be left to the discretion of the systems integrator. Only functional requirements are specified in this document, the amount of devices it takes to accomplish each function may be variable.

4.1. (NU) FUNCTIONAL OVERVIEW

This section will describe the functional areas of the SBS and how they relate to the standards provided in Annex A-J.

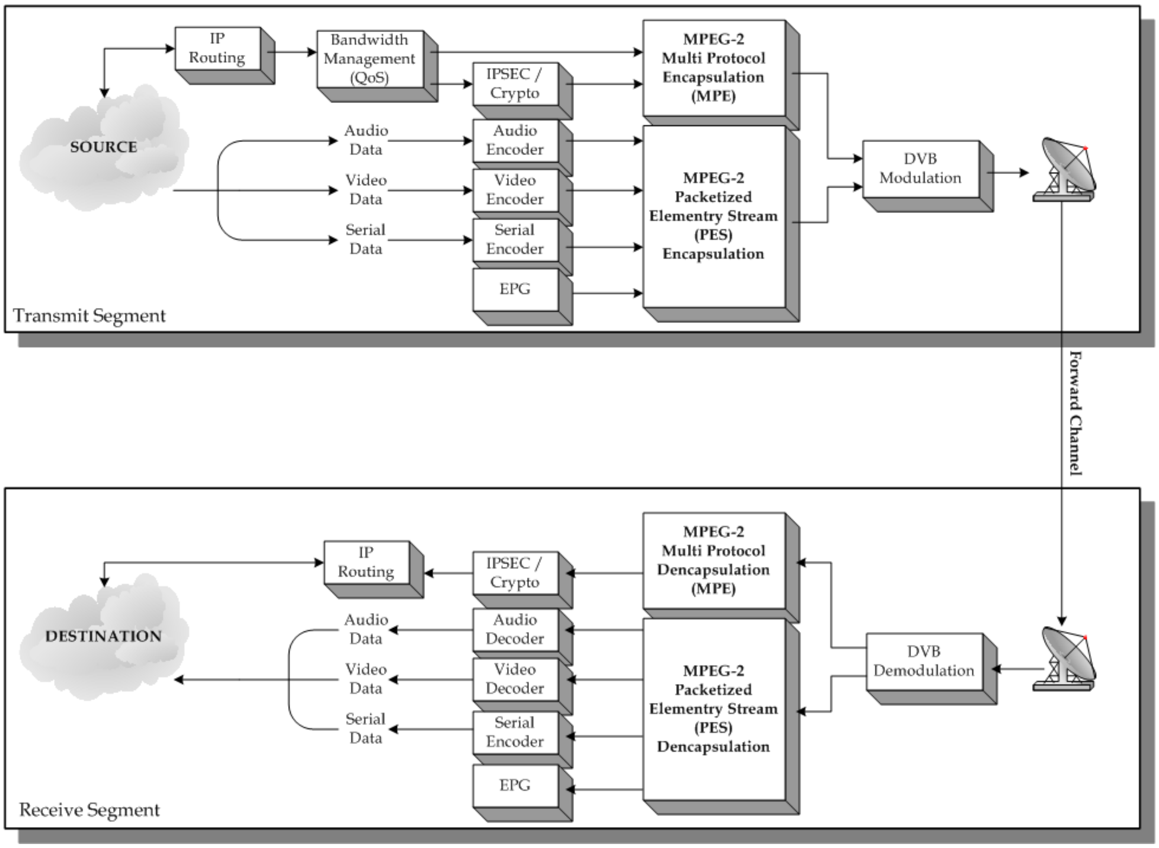

Figure 1 - SBS Functional Overview

Figure 1 depicts each of the functional areas within the transmit and receive segments. Because all receive functions are operated in reverse of the transmission gateway functions, the specifications in Annex A-J do not provide separate definitions for the receive gateway. It is assumed that systems integrators and vendors will carry out the converse order of operations for all receive gateway functions.

4.1.1. (NU) Routing and Traffic Management

The transmit segment can be made flexible in the types of information that is incorporated into the broadcast, meaning different types of data may come off of a variety of networks (i.e., IP, ATM, POS, etc.). The mainstay requirement for the transmission gateway is to have all data in IP format prior to MPEG-2 encapsulation and ultimately DVB modulation. The transmit gateway shall broadcast network data in IP format only. The STANAG assumes that all measures will be taken in order to ensure network data being broadcast is in IP format and therefore functions used to interoperate with data source networks other than IP are not specified. The SBS transmit gateway shall be required to support Border Gateway Protocol Version 4 (BGP-4) as its Exterior Gateway Protocol (EGP) for IP networking. Specific definitions for BGP-4 can be found in Annex B.

Data can be passed through the transmit gateway as unicast or multicast IP traffic. Unicast traffic delivery is straightforward. However, multicast traffic flow requires gateway components to statically join multicast groups or dynamically join groups via Internet Group Membership Protocol (IGMP). IGMP can be employed in two manners: 1. IGMP use can be isolated to the transmit and receive segment networks. In this instance, the transmit and receive segment components will join pre-arranged multicast groups independently. 2. IGMP can be deployed in an end-to-end configuration if a return channel is present. This causes transmit gateway components to join multicast groups as requested by the receive segment components. Return channel implementations using the DVB-Return Channel Satellite (DVB-RCS) are described in Annex I. Annex B provides the definition for IGMP. Implementation of this protocol is left to the discretion of the systems integrator. (Multicast routing may not be enabled on a network or may not be supported by some cryptographic devices. For these cases, Annex B provides a definition for IP Tunneling, which can be used as a work around for this type of situation).

As data is received from the source network there are various techniques that may be used to route the data internally within the transmit segment. These implementations are not critical to interoperability so long as IP addressing and Packet Identifier (PID) information are exchanged or kept at a universally known constant. IP routes within the transmit segment can be static, default, or dynamic routes depending upon the architecture and applications being employed. Annex B provides definition for IP routing requirements.

The transmit segment shall have the capability to allocate bandwidth based on IP Type of Service, IP address, and protocol (port). This is necessary to prevent a single data source from utilizing more than its share of the total system bandwidth. Annex C provides the definition for traffic shaping.

4.1.2. (NU) Audio/Video and Serial Stream Data

Analog A/V and Synchronous Serial Stream supportability is not a mandatory requirement for NATO SBS. However, if these services are going to be provided on any NATO SBS system it shall be done in compliance with the definitions referenced in Annex E & F of this STANAG.

4.1.3. (NU) MPEG-2 Encapsulation

IP, Analog Audio/Video, and Serial Stream data shall be encapsulated into the MPEG-2 Transport Stream prior to DVB modulation. IP data encapsulation will follow the Multi Protocol Encapsulation (MPE) specification provided in Annex D. Analog A/V and Serial data encapsulation (if implemented) will follow the Packetized Elementary Stream (PES) encapsulation specifications referenced in Annexes E&F.

4.1.4. (NU) Digital Video Broadcast (DVB) Modulation

The transmit gateway may be used with multiple frequency bands to operate with a variety of space segments. Two standards based on the Digital Video Broadcast (DVB) standard are described in this document. Annex H describes the DVB over satellite (DVB-S) waveform coding and modulation standard. Annex J describes the second generation DVB over satellite (DVB-S2) waveform coding and modulation standard. Each standard describes the step-by-step processing from the multiplexing of data up to the transmitting antenna. Annex H and Annex J provide guidance to ensure systems maintain a minimal set of interoperable functions, leaving the choice of standard to implement at the discretion of the system integrator.

4.1.5. (NU) Interactive Services Return Channel

The SBS may implement an optional return channel to facilitate interactive services to the receive segment. This return channel will allow receive segment terminals to transmit management and control (M&C) and data traffic back to the gateway, enabling functions such as deployed injection, end-to-end dynamic IP routing, and link feedback for adaptive coding and modulation (ACM). Annex I describes the application of the DVB-RCS standard with required and recommended supplemental functions that foster interoperability in the SBS.

5. (NU) IMPLEMENTATION OF THIS AGREEMENT

Nations compliance to this agreement shall be satisfied when SBS gateway in that nation's forces are placed in service and interoperate with other nations' SBS gateways that comply with the characteristics detailed in this STANAG.

ANNEX A: TERMS AND DEFINITIONS

6. (NU) ACRONYMS

The following acronyms are used for the purpose of this agreement.

8PSK	8-ary Phase Shift Keying
16APSK	16-ary Phase Shift Keying
32APSK	32-ary Phase Shift Keying
ACM	Adaptive Coding and Modulation
ACQ	Acquisition
AS	Autonomous System
ATM	Asynchronous Transfer Mode
AVBDC	Absolute Volume Based Dynamic Capacity
BBHEADER	Baseband Header
BCH	Bose Chaudhuri Hocquenghem
BGP-4	Border Gateway Protocol version 4
CAS	Conditional Access System
CAT	Conditional Access Table
CBR	Constant Bit Rate
CCM	Constant Coding and Modulation
CMT	Correction Message Table
CRA	Continuous Rate Assignment
CRC	Cyclic Redundancy Check
CSC	Common Signaling Channel
DARPA	Defense Advanced Research Projects Agency (US DoD)
DSM-CC	Digital Storage Media – Command and Control
DULM	Data Unit Labeling Method
DVB	Digital Video Broadcasting
DVB-RCS	Digital Video Broadcasting – Return Channel Satellite
DVB-S	Digital Video Broadcast over Satellite
DVB-S2	Digital Video Broadcast over Satellite, second generation
ECM	Entitlement Control Message
EGP	Exterior Gateway Protocol
EIGRP	Enhanced Interior Gateway Routing Protocol
EMM	Entitlement Management Message
EN	European Norm (ETSI)
ESP	Encapsulating Security Payload
ETSI	European Telecommunications Standards Institute
FCA	Free Capacity Assignment
FCT	Frame Composition Table
FEC	Forward Error Correction
FIFO	First In First Out

GHz	Gigahertz
GRE	Generic Routing Encapsulation
HAIPE	High Assurance Internet Protocol Encryption
HAIPIS	High Assurance Internet Protocol Interoperability Specification
HDTV	High Definition Television
HP	High Priority
Hz	Hertz
IANA	Internet Assigned Numbers Authority
ICV	Integrity Check Value
IETF	Internet Engineering Task Force
IGMP	Internet Group Membership Protocol
IP	Internet Protocol
IRD	Integrated Receiver Decoder
ISS	Input Stream Synchronizer
ITU	International Telecommunications Union
KHz	Kilohertz
LAN	Local Area Network
LDPC	Low Density Parity Check
LLC	Logical Link Control
LP	Low Priority
M&C	Management and Control
MAC	Medium Access Control
MAS	Military Agency for Standardization
MBGP	Multiprotocol Border Gateway Protocol
MF TDMA	Multiple Frequency Time Division Multiple Access
MMT	Multicast Mapping Table
MPE	Multi-Protocol Encapsulation
MPEG	Moving Pictures Expert Group
MUX	Multiplexer
NATO	North Atlantic Treaty Organization
NCC	Network Control Center
NCR	Network Clock Reference
OSPF	Open Shortest Path First
PCR	Program Clock Reference
PES	Packetized Elementary Stream
PID	Packet Identifier
PIM	Protocol Independent Multicast
PLFRAME	Physical Layer Frame
PLS	Physical Layer Signaling
PRBS	Pseudo Random Binary Sequence
PSI	Program Specific Information
QoS	Quality of Service

QPSK	Quadrature Phase Shift Keying
RBDC	Rate Based Dynamic Capacity
RCST	Return Channel Satellite Terminal
RFC	Request For Comment (IETF)
RIB	Routing Information Base
RIP	Routing Information Protocol
RMT	RCS Map Table
RS	Reed Solomon
SA	Security Association (HAIPIS)
SAC	Satellite Access Control
SBS	Satellite Broadcast Service
SDTV	Standard Definition Television
SI	Service Information
SIT	Satellite Interactive Terminal
SNAP	Sub Network Access Protocol
SOF	Start Of Frame
SPI	Security Parameter Index (ESP)
SPT	Satellite Position Table
STANAG	Standardization Agreement (NATO)
SYNC	Synchronization
SYNCD	Synchronization Distance
TBD	To Be Determined
TBTP	Terminal Burst Time Plan
TCP	Transport Control Protocol
TCT	Timeslot Composition Table
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TMST	Transmission Mode Support Table
TR	Technical Report (ETSI)
TRF	Traffic
TS	Transport Stream (MPEG)
UDP	User Datagram Protocol
UP	User Packet
VBDC	Volume Based Dynamic Capacity
VBR	Variable Bit Rate
VCM	Variable Coding and Modulation
VPN	Virtual Private Networking
WAN	Wide Area Network

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ANNEX B: IP ROUTING

1. (NU) INTRODUCTION

This annex describes the minimum set of IP protocols to be employed by NATO SBS in order to support IP data transmissions across the satellite. NATO SBS shall support Unicast and Multicast IP data flow.

1.1. (NU) AIM

This annex describes the IP protocol mechanisms to be used by NATO SBS.

1.2. (NU) GENERAL

Figure B - 1 highlights the portion of the SBS that shall be covered by this annex.

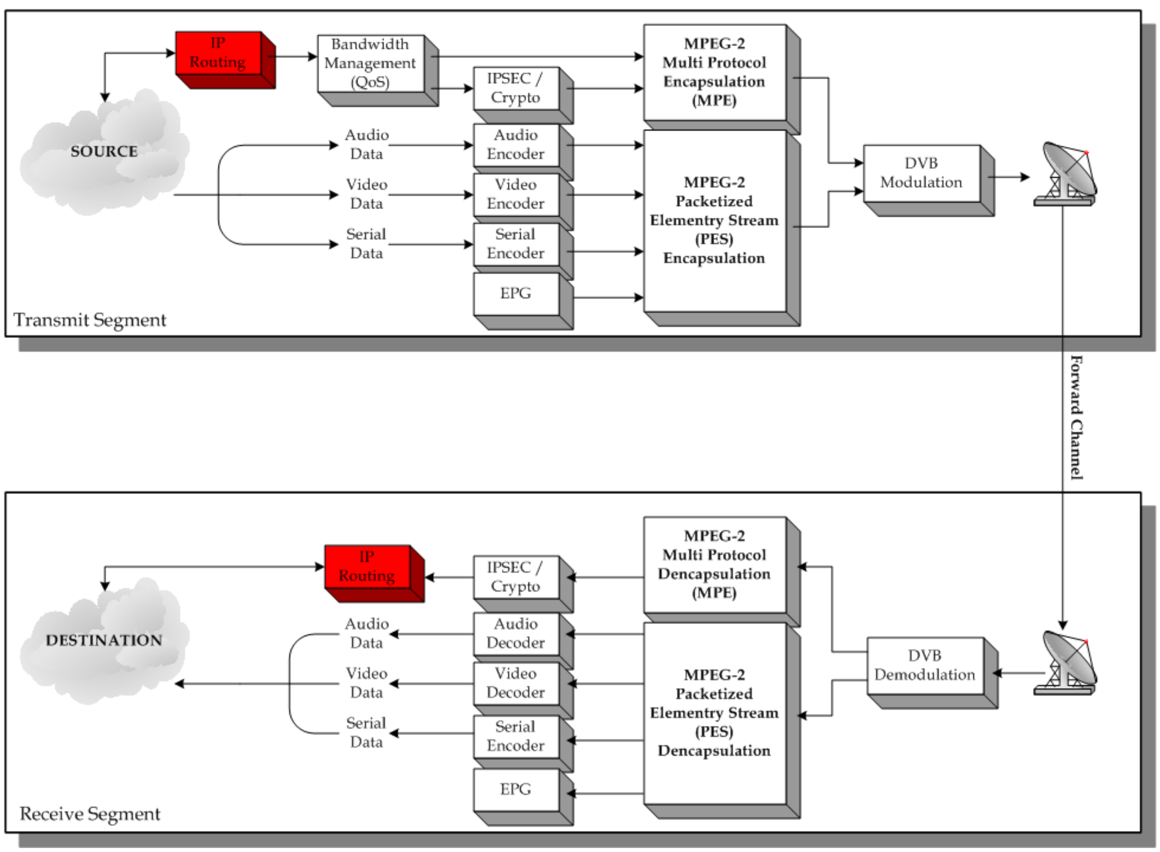


Figure B - 1 IP Routing Functional Location

2. (NU) UNICAST DATA FLOW

NATO SBS shall support two types of transport layer protocols for unicast data transmission: User Data Protocol (UDP) and Transmission Control Protocol (TCP). NATO SBS shall be compliant with RFC 768 and RFC 793.

2.1. (NU) USER DATAGRAM PROTOCOL (UDP)

UDP is a connection-less transport layer protocol. It provides faster data transfers and has less overhead than TCP because it lacks a reliability mechanism. More information on UDP can be obtained from RFC 768. The syntax of a UDP segment is illustrated in Figure B - 2.

<table>
<tr><td>0</td><td>15</td><td>31</td></tr>
<tr><td colspan="2">Source Port</td><td>Destination Port</td></tr>
<tr><td colspan="2">Length</td><td>Checksum</td></tr>
<tr><td colspan="3">Data (variable)</td></tr>
</table>

The source port is an optional field, which indicates the port number of the sending application. The length field indicates the length of the entire packet, including the header. The 2-byte checksum is a one's complement of the one's complement sum of source address, destination address, protocol, and UDP length.

2.2. TRANSMISSION CONTROL PROTOCOL

TCP is a connection-oriented transport layer protocol and will be used in duplex environments where data transport reliability is required. More information on TCP can be obtained from RFC 793. The syntax of the TCP header is illustrated in Figure B - 3. Table B - 1 summarizes the semantics of each of the TCP header fields.

<table>
<tr><td colspan="5">Source Port</td><td colspan="5">Destination Port</td></tr>
<tr><td colspan="10">Sequence Number</td></tr>
<tr><td colspan="10">Acknowledgment Number</td></tr>
<tr><td>Data
Offset</td><td>Reserved</td><td>U
R
G</td><td>A
C
K</td><td>P
S
H</td><td>R
S
T</td><td>S
Y
N</td><td>F
I
N</td><td colspan="2">Window</td></tr>
<tr><td colspan="8">Checksum</td><td colspan="2">Urgent Pointer</td></tr>
<tr><td colspan="9">Options</td><td>Padding</td></tr>
</table>

Figure B - 3 TCP Header

Table B - 1 Semantics of TCP header fields

Field	Length	Description
Source Port	16 bits	Source Port Number
Destination Port	16 bits	Destination Port Number
Sequence Number	32 bits	Sequence number of first data octet in the segment if SYN is not present; Initial sequence number when SYN is present
Acknowledgement Number	32 bits	When ACK is 1, value of next sequence number sender is expecting to receive
Data Offset	4 bits	Number of words in TCP header
Reserved	6 bits	Must be zero
URG	1 bit	Urgent Pointer field Significant (control bit)
ACK	1 bit	Acknowledgement field significant (control bit)
PSH	1 bit	Push Function (control bit)
RST	1 bit	Reset the Connection (control bit)
SYN	1 bit	Synchronize sequence numbers (control bit)
FIN	1 bit	No more data from sender (control bit)
Window	16 bits	Number of bytes sender is willing to accept
Checksum	16 bits	One's complement of one's complement sum of all 16 words in header and text
Urgent Pointer	16 bits	Value of urgent pointer as offset from sequence number
Options	Variable	Examples include End of Option List, No-operation, Maximum Segment Size, and Maximum Segment Size Option Data
Padding	Variable	To ensure TCP header is 32 bits long

2.3. (NU) STATIC IP ROUTING

NATO SBS shall provide the capability to employ static IP routes. Static routing is the most basic solution for programming an IP routing device to transfer inbound data to a specified outbound interface.

2.4. (NU) DYNAMIC IP ROUTING

NATO SBS shall support dynamic IP routing protocols as required by external network interfaces and the Asymmetrical Networking option, if implemented. Interior routing protocols are not specified in this document (e.g., OSPF, RIP, EIGRP, etc); it is assumed that all devices will be equipped with the appropriate interfaces and protocols needed to communicate with the local infrastructure.

2.4.1. (NU) Border Gateway Protocol Version 4 (BGP-4)

NATO SBS shall support the Border Gateway Protocol (BGP) Version 4, as defined in RFC 1771. BGP-4 is a standard TCP/IP routing protocol intended to provide routing capability between two autonomous networks. BGP-4 is a path vector protocol that exchanges routing updates between peers over a TCP connection. BPG-4 supports classless interdomain routing, as well as aggregation of routes and Autonomous Systems (AS). An Autonomous System is a set of routing devices under a single administration, using a common interior gateway protocol and common routing techniques for routing within the AS, and an exterior gateway protocol to route packets to other ASs.

BGP-4 will need to be supported on the local terrestrial network that the SBS hub and remotes will interface with. The SBS will need to be able to interface between the interior and exterior routing protocols (OSPF, RIP, EIGRP) as motioned in section 2.4

2.4.1.1. (NU) Routing Information Bases (RIB)

Routes are stored in the Routing Information Bases (RIBs). RIBs are comprised of three parts: Adj-RIBs-In, Loc-RIB, and Adj-RIBs-Out. The Adj-RIBs-In store unprocessed information obtained from inbound update messages. Loc-RIB contains local routing information, which is obtained by applying local policies to the inbound information stored in the Adj-RIBs-In. The Adj-RIBs-Out stores the information that the local BGP speaker advertises to its peers.

2.4.1.2. (NU) Message Syntax

Each message has a fixed header length of 19 bytes. Figure B - 4 depicts the syntax of the message header. Data may or may not follow the header, depending on the message type.

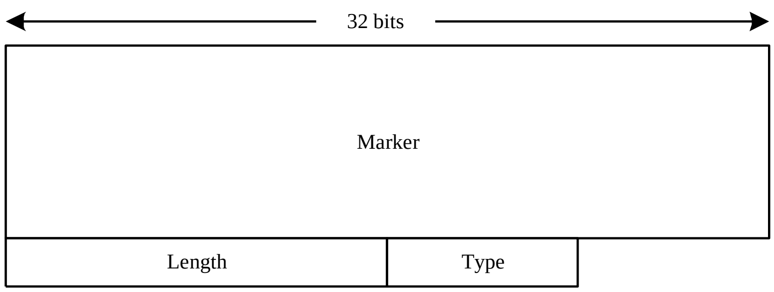


Figure B - 4 BGP-4 Message Header

The marker is a 16-byte field contains a predictable value. If the message is open or if the open message contains no Authentication Information, then the Marker will consist of all ones. Otherwise, the value of the marker can be computed by a authentication mechanism. Some functions of this field include synchronization loss detection and authentication of incoming messages.

The length of the message, defined in the two-byte length field, must be between 19 and 4096. The one-byte type field defines the message type. Table B - 2 depicts the four types that are defined in RFC 1771.

Table B - 2 Message Types for BGP-4

Value	Type Code
1	OPEN
2	UPDATE
3	NOTIFICATION
4	KEEPALIVE

Once a transport protocol connection has been made, an OPEN message will be sent to each side. If these messages are received, a KEEPALIVE message is sent to confirm the receipt. After this process has successfully occurred, UPDATE, KEEPALIVE, and NOTIFICATION messages may be sent for the duration of the connection.

UPDATE messages transfer routing information. It can be used to advertise a single feasible route to a peer or withdraw unfeasible routes. The KEEPALIVE message contains no data besides the message header. Sending KEEPALIVE messages prevents the Hold Timer from expiring. It is recommended that KEEPALIVE messages be sent at an interval one third of the Hold Time interval. NOTIFICATION messages are sent when an error is detected in the transmission of any BGP message.

Figure B - 5 depicts the syntax for each of these messages.

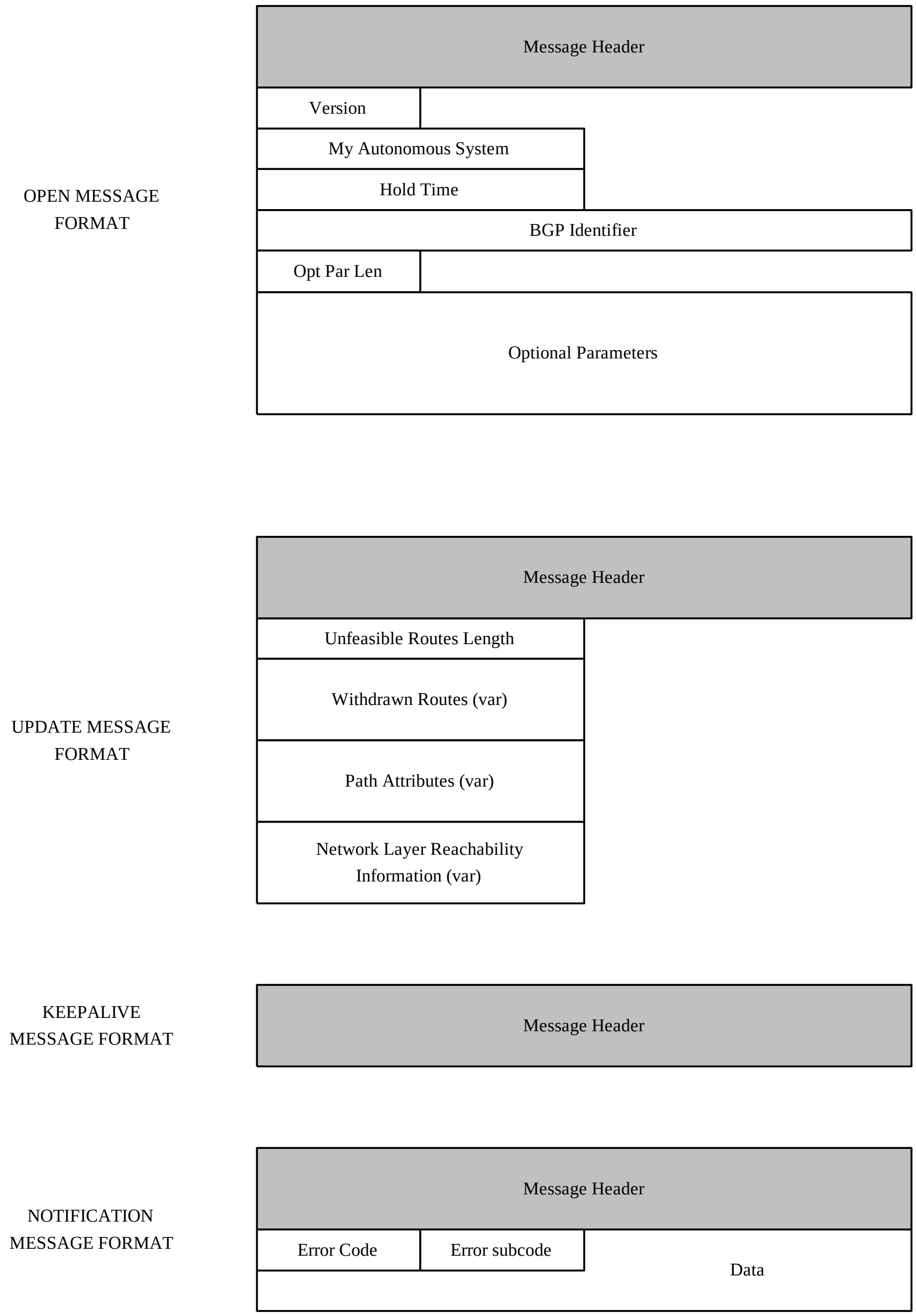


Figure B - 5 BGP-4 Message Formats

2.4.1.3. (NU) UPDATE Messages

There are four types of path attributes: well-known mandatory, well-known discretionary, optional transitive, and optional non-transitive. Well-known attributes are understood by all BGP implementations: the mandatory attributes must be included in all UPDATE messages, whereas discretionary do not. Not all BGP implementations support the optional attributes. If the transitive bit is set, the transitive optional attributes will be accepted. Non-transitive optional attributes that are not recognized will be ignored. For more details on various attributes, refer to RFC 1771.

2.4.1.4. (NU) Error Handling

When an error occurs in the processing of BGP messages, a NOTIFICATION message is sent and the BGP connection is closed. If there is an error with the delivery of a NOTIFICATION message, there is no means of reporting this error.

3. MULITCAST DATA FLOWS

NATO SBS shall provide IP Multicast services. NATO SBS shall be compliant with RFC 2236, 1771, and 2283. This sections provides a description of these RFCs.

3.1. (NU) STATIC MULTICAST GROUP MEMBERSHIP

NATO SBS shall provide the capability to manually join/leave multicast groups of interest.

3.2. (NU) INTERNET GROUP MANAGEMENT PROTOCOL (IGMP)

IGMP allows for the creation and maintenance of memberships to neighboring multicast routing devices. This section will reference the RFC 2236, which is the standard for IGMP Version 2. The Internet Draft IGMP-V3 proposes the standard for Version 3 of IGMP.

Multicast routing devices use IGMP to determine group memberships for each of their physical networks. The router maintains a membership list and will either be a Querier or Non-Querier on each of its attached networks. There will be only one Querier per physical network. If a Query message is sent from another router with a lower IP address, the receiving router must change roles to a Non-Querier.

When a host joins a group, it will transmit a Membership report for that group. When a host leaves the group, it will transmit a Leave Group message.

The three types of router-interaction messages include:

- Membership Query (0x11) – General Queries are used to learn which groups have memberships for attached networks. Group-specific queries are used to learn if a specific group has any members on an attached network
- Version 2 Membership Report (0x16)
- Leave Group (0x17)

IGMP messages are encapsulated within IP packets, with the protocol number equal to 2 and the Time-To-Live (TTL) equal to one. Figure B - 6 illustrates the format of an IGMP message.

Type	Max Resp Time	Checksum
Group Address		

Figure B - 6 IGMP Message Format

3.2.1. (NU) IGMP-Proxying

IGMP-based Multicast Forwarding can be utilized in certain situations rather than implementing a multicast routing protocol. The following sections will describe this process, in accordance with the IGMP-Proxy-00 Internet Draft.

The remote terminal performing IGMP-Proxying will have one upstream interface and multiple downstream interfaces. The terminal does not send a membership report that is produced by a user located on its network interface. The Terminal needs additional information than the one provided by the IGMP messaging to be able to receive a multicast flow. One of them is the PID number that is use to transport a given “multicast flow.” The “Multicast Mapping Table (MMT)” is defined in Annex I to provide such information

3.3. (NU) STATIC IP MULTICAST ROUTING

NATO SBS shall provide the capability to employ static IP multicast routes. Static routing is the most basic solution for programming an IP routing device to transfer inbound data to a specified outbound interface.

3.3.1. Protocol Independent Multicast, Version 2 (PIMv2)

NATO SBS shall be compliant with ratified Internet RFC for Protocol-Independent Multicast (PIM) Dense Mode & Sparse Mode - Protocol Specifications. PIM works with all existing unicast routing protocols. It supports two different types of multipoint traffic distribution patterns: Dense and Sparse.

Dense mode is most useful when:

- Senders and receivers are in close proximity to one another.
- There are few senders and many receivers.
- The volume of multicast traffic is high.
- The stream of multicast traffic is constant.

Sparse multicast is most useful when:

- There are few receivers in a group.
- Senders and receivers are separated by WAN links.
- The type of traffic is intermittent.

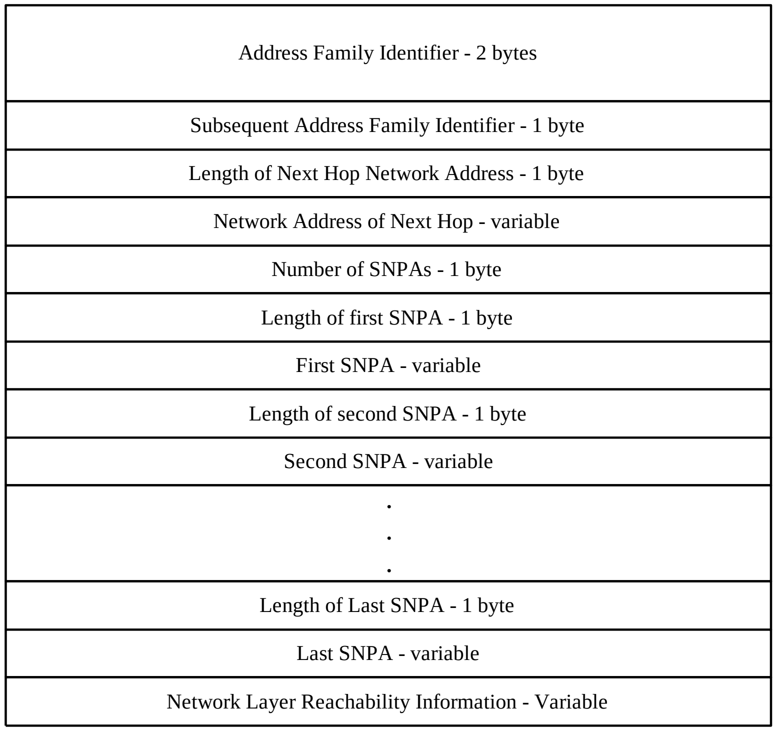
PIM is able to simultaneously support dense mode for some multipoint groups and sparse mode for others.

3.3.2. (NU) Multiprotocol Extensions for BGP (MBGP)

NATO SBS shall be compliant with RFC 2283, which defines Multiprotocol BGP (MBGP). Much like BGP-4 the back-end network of the Hub will need to support MBGP. MBGP is an enhanced BGP that carries two sets of routes, one set for unicast routing and one set for multicast routing. MBGP provides the capability to connect multicast topologies within and between Autonomous Systems. Multiprotocol BGP routes cannot be redistributed into BGP.

The three fields carried by BGP-4 that are IPv4-specific are NEXT_HOP, AGGREGATOR, and NLRI. In order for BGP-4 to support routing to multiple network layer protocols, BGP-4 must have the ability to associate the network layer protocol with the next hop information and with NLRI. RFC 2283 specifies two attributes to be added to MBGP: Multiprotocol Reachable NLRI (MP_REACH_NLRI) and Multiprotocol Unreachable NLRI (MP_UNREACH_NLRI). These attributes are used to carry the set of reachable and unreachable destinations, respectively. MP_REACH_NLRI is used to advertise feasible routes. The syntax for this attribute is shown in Figure B - 7.

Figure B - 7 Syntax for the MP_REACH_NLRI field



The MP_UNREACH_NLRI attribute may be used to discard unfeasible routes. Figure B - 8 depicts the syntax of this attribute.

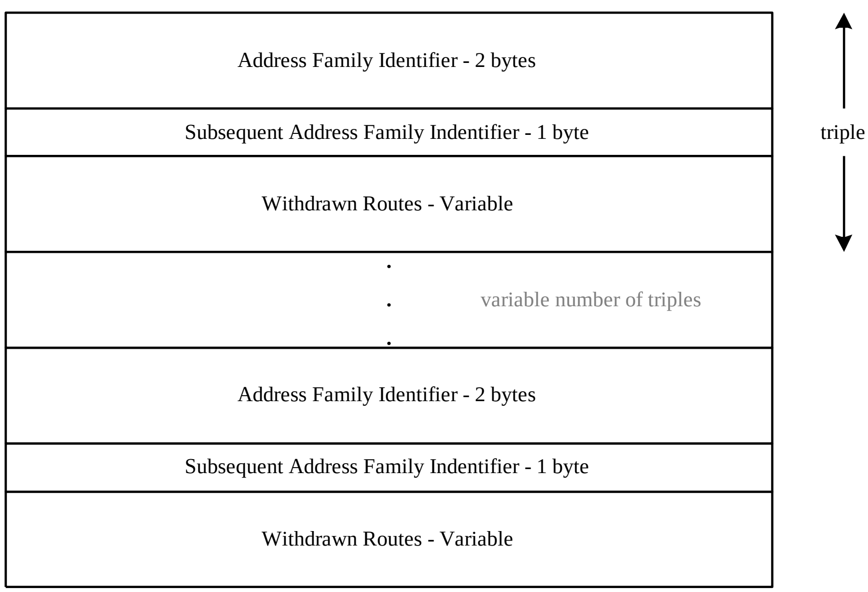


Figure B - 8 Syntax of MP_UNREACH_NLRI field

4. IP TUNNELING

NATO SBS shall be compliant with RFC 2784 and RFC 3077 in order to provide mechanisms for IP-in-IP tunneling. This is necessary to support transmissions across non-multicast enabled networks as well as non-multicast enabled encryption devices. Tunneling can be implemented between the Hub router and the Terminal router, but is not limited between hub and terminal. It can be extended to the networks on either side.

4.1. (NU) GENERIC ROUTING ENCAPSULATION (GRE)

Generic Routing Encapsulation (GRE) provides the capability to encapsulate one protocol over another as specified in RFC 2784. Payloads will be GRE encapsulated prior to encapsulation by the transport protocol. Figure B - 9 illustrates the syntax of a GRE packet. When IP packets are being encapsulated, The Protocol type will be set to 0x800.

<table>
<tr><td>CS
Pres</td><td>Reserved</td><td>Version</td><td>Protocol Type</td></tr>
<tr><td colspan="3">Checksum (optional)</td><td>Reserved1 (optional)</td></tr>
<tr><td colspan="4">payload packet</td></tr>
</table>

Figure B - 9 Syntax of a GRE packet

The first bit, checksum present, will determine the presence of the optional checksum. The reserved fields must be all zeros. All packets received that have non-zero values in these fields will be ignored.

ANNEX C: QUALITY OF SERVICE (QOS)

1. INTRODUCTION

NATO SBS shall implement Quality of Service (QoS) mechanisms to ensure bandwidth can be allocated based on the Type of Service, IP address, and protocol (port).

1.1. (NU) AIM

This annex describes the requirement for SBS to support an IP QoS mechanism.

1.2. (NU) GENERAL

Figure C - 1 highlights the portion of the SBS that shall be covered by this annex.

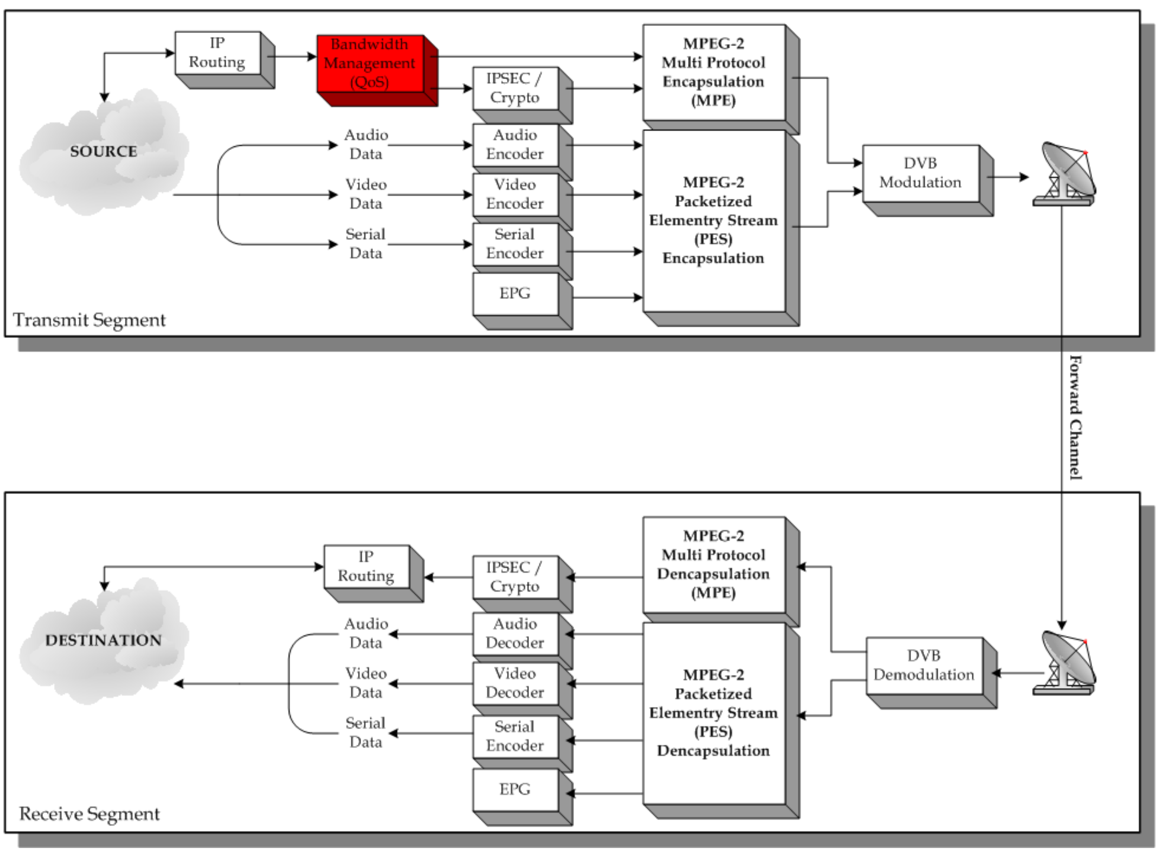


Figure C - 1 QoS Functional Location

2. (NU) QOS

NATO SBS requires a Quality of Service (QoS) mechanism to ensure different types of data are guaranteed a specific amount of bandwidth over the broadcast. This is needed to prevent a single type of data transmission from flooding the link. The system shall have the capability to allocate bandwidth based on IP address, type of service, and protocol (port).

ANNEX D: IP DATAGRAM ENCAPSULATION

1. (NU) INTRODUCTION

This annex describes the process used to encapsulate Internet Protocol (IP) data into MPEG-2 Transport Streams (ref. ETSI EN 301 192, ISO/IEC 13818-1 & 13818-6) as required to support DVB Modulation and Channel Coding for NATO SBS (ref. ETSI EN 300 421). DVB Data Broadcasting is to be implemented in conjunction with the DVB specification for the format and implementation of Service Information (SI) provided in EN 300 468 and ETR 211.

1.1. (NU) AIM

This is an outline of relevant specifications provided in ETSI EN 301 192, ISO/IEC 13818-1, and ISO/IEC 13818-6 for DVB Multiprotocol Encapsulation (MPE).

1.2. (NU) GENERAL

Figure D - 1 highlights the portions of the SBS that are covered in this annex.

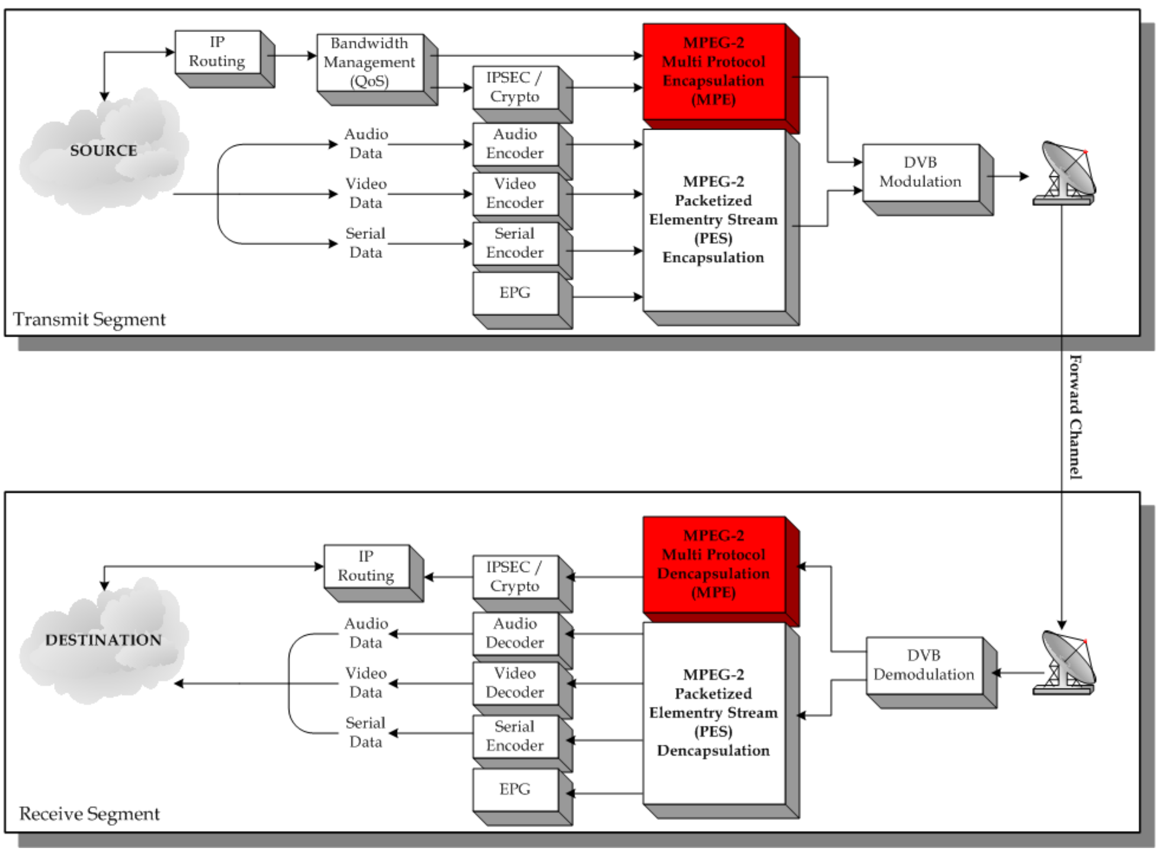


Figure D - 1 MPE Functional Location

1.2.1. (NU) Internet Protocol

The SBS system shall provide the appropriate interfaces needed to accept IP data as an input to the system or format data that is input to the system as IP datagrams. The IP protocol is a connectionless, independently routed protocol used to transmit datagrams across IP-enabled internetworks. IP is the standard network protocol used for Internet communications. For more information on the Internet Protocol standard, refer to RFC 791: Internet Protocol – DARPA Internet Program Protocol Specification.

1.2.1.1. (NU) Internet Header Format

The format of the Internet Protocol version 4 header (IPv4) is illustrated in Figure D - 2. The format of the Internet Protocol version 6 (IPv6) header is illustrated in Figure D - 3

The SBS will support IPv4 as the standard. IPv6 will be supported as a future option as it becomes readily available and implemented into the architecture.

Version	IHL	Type of Service	Total Length
Identification		Flags	Fragment Offset
Time to Live		Protocol	Header Checksum
Source Address			
Destination Address			
Options			Padding

Figure D - 2 Internet Protocol Header

0	4	12	16	24	31
Version	Class	Flow Label			
Payload Length			Next Header	Hop Limit	
Source Address (128bit)					
Destination Address (128bit)					

Figure D - 3 Internet Protocol Header

2. (NU) MULTIPROTOCOL ENCAPSULATION (MPE)

IP datagrams shall be encapsulated into MPEG-2 Transport Streams in accordance with:

- the datagram_sections format of EN 301 192
- the DSMCC_Section format of ISO/IEC 13818-6
- the private_section semantics of ISO/IEC 13818-1.

This encapsulation is described below.

2.1. (NU) DATAGRAM_SECTION

For MPEG-2 Systems, ISO/IEC 13818-1 defines a private_section structure which enables private information to be carried within the MPEG-2 transport stream packets. Private data is defined as being any user data that is not coded according to a standard specified by the ISO/IEC 13818-1 Specification. The private_section provides the minimum structure required for a compliant MPEG-2 decoder to parse the data stream.

In accordance with EN 301 192, the transmission of datagrams (IP Packets) is accomplished by encapsulating the datagrams into datagram_sections. Datagram_sections are compliant to the DSMCC_Section format described in ISO/IEC 13818-6 and are mapped into the MPEG-2 transport stream in compliance with the private_section semantics defined in ISO/IEC 13818-1. The DSMCC_Section format includes all of the private_section syntax so that compliant MPEG-2 System decoders may be used. The syntax for datagram_sections is shown in Figure D - 4.

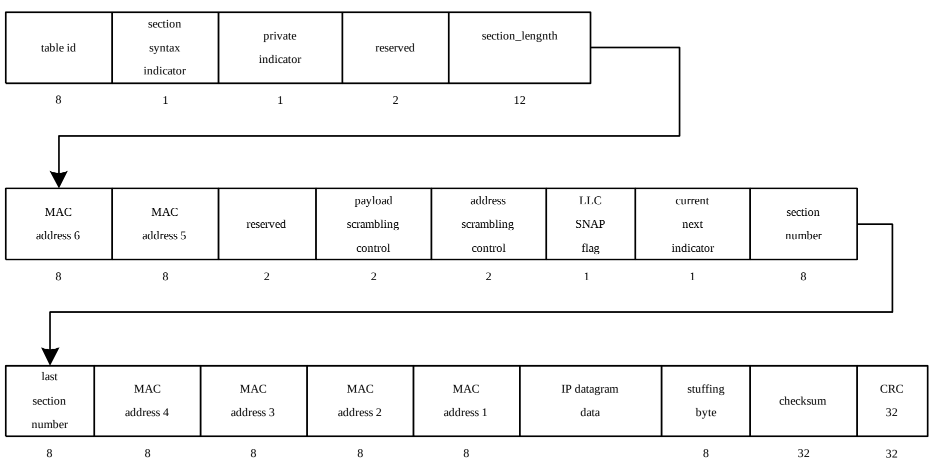


Figure D - 4 Datagram_section syntax

The table_id field of the datagram_section will be 0x3E, which indicates DSMCC_sections containing private data. IP datagrams will be segmented and encapsulated as private data. The reserved field will be set to '11'.

The MAC_address_[1.6] fields will contain the MAC address of the destination. This destination address will be fragmented into six eight-bit fields, corresponding to MAC_address _1, containing the most significant bit, through MAC_address_6, containing the least significant bit. Figure D - 5 depicts the mapping of the MAC destination address field.

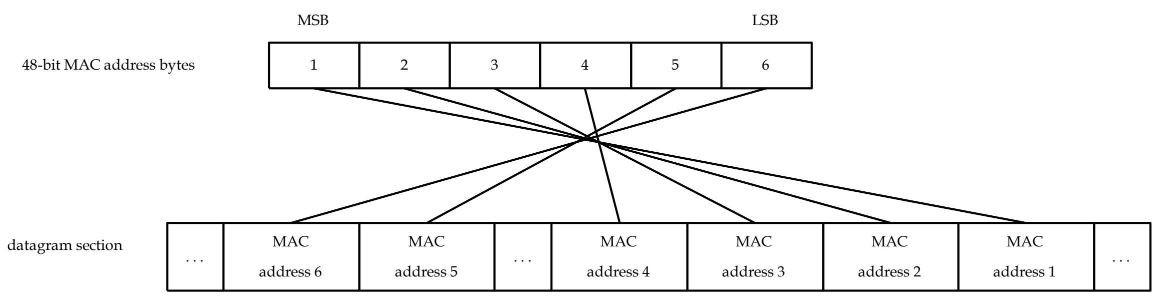


Figure D - 5 Mapping of MAC address to datagram_section

The LLC_SNAP_flag will be set to one if the payload carries an LLC/SNAP encapsulated datagram. This structure will indicate the type of datagram being encapsulated. The current_next_indicator will be set to one. The 8-bit section_number indicates the portion of the datagram being carried in the case where the datagram spans multiple sections. If the datagram spans multiple sections, the last_section_number will indicate the last section, which will carry the datagram. The LLC_SNAP structure will comply with the ISO/IEC 8802-2 Logical Link Control and ISO/IEC 8802-1 Subnetwork Attachment Point specifications.

The IP_datagram_byte will contain the bytes of the datagram. The checksum will be calculated over the entire datagram_section. Performing one's complement addition over the 32 bits of the datagram_section, and then taking the one's complement of the result, as defined in ISO/IEC 13818-6, will calculate the checksum. The CRC_32 field is defined in ISO/IEC 13818-1 Annex B, and is only present when the section_syntax field is equal to one.

2.1.1. (NU) Program Specific Information (PSI) and Service Information (SI)

The data_broadcast _descriptor described in EN 301 192 will be used to indicate transmission of datagrams. The stream type will be indicated in the program map section described in ISO/ESI 13818-1 by setting the value of the stream type to 0x0D.

2.2. (NU) DATAGRAM_SECTION ENCAPSULATION INTO MPEG-2 TRANSPORT STREAM

Transport Steam packets have a fixed length of 188 bytes, including null packets and stuffing bytes. ISO/IEC 13818-1 defines a mechanism that allows several MPE packets to be placed in the same MPEG-2 packet using the pointer field. This mechanism is known as section packing. The SBS will implement Section Packing on the forward link

2.2.1. (NU) MPEG-2 Transport Stream Header Syntax

Figure D - 6 depicts the Transport Stream Header format.

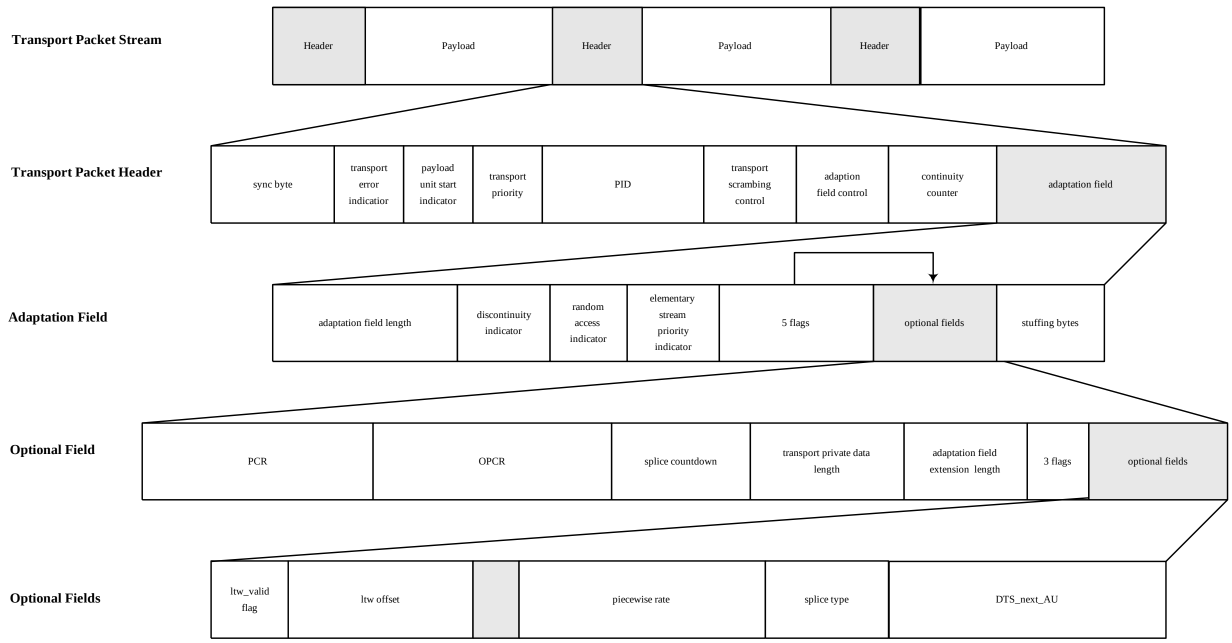


Figure D - 6 Transport Stream Header

2.2.2. (NU) Transport Stream Payload Syntax

Figure D - 7 Illustrates the IP datagram encapsulation into the Transport Stream payload format.

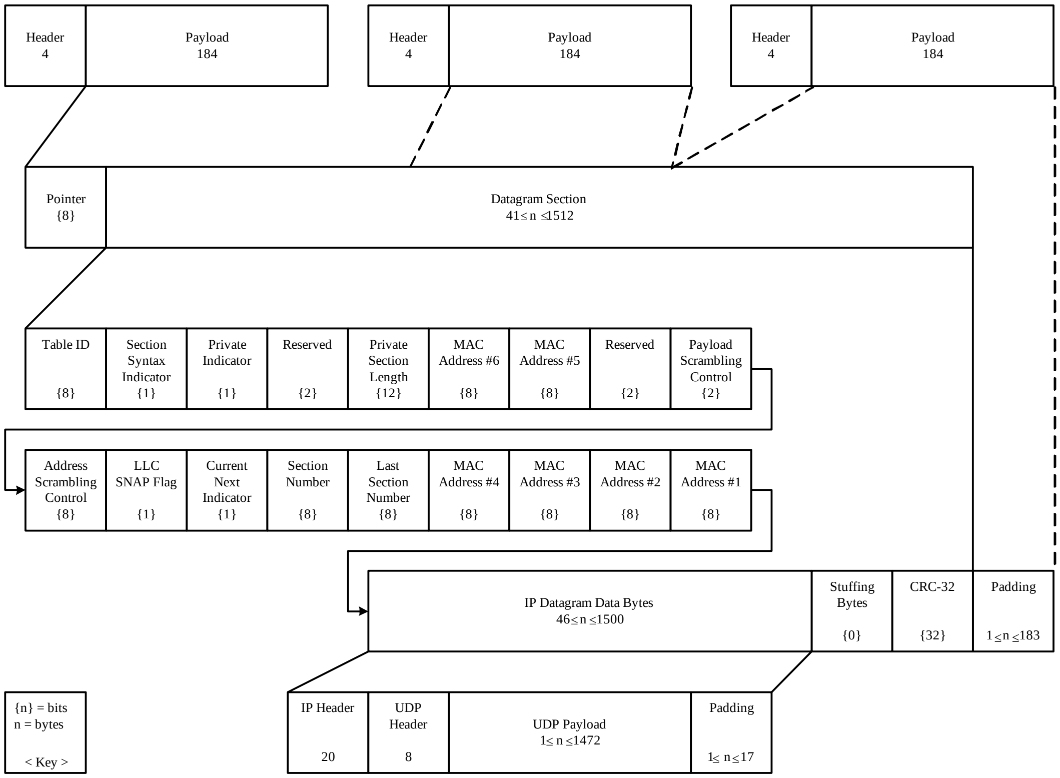


Figure D - 7 IP Datagram Encapsulation Into MPEG-2 Transport Stream

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ANNEX E: SYNCHRONOUS SERIAL STREAM ENCAPSULATION

1. (NU) INTRODUCTION

This annex describes the process used to deliver synchronous serial data via MPEG-2 Transport Streams (ref. ETSI EN 301 192, ISO/IEC 13818-1) as required to support DVB Modulation and Channel Coding for NATO SBS (ref. ETSI EN 300 421). Synchronous data is defined as streaming data with timing requirements such that the data and clock signal can be regenerated at the receiver into a synchronous serial stream (e.g., E1, T1).

Synchronous serial stream supportability is not a mandatory requirement for NATO SBS. The information provided herein is required only if the option to support synchronous serial stream is implemented.

1.1. (NU) AIM

This is an outline of relevant specifications provided in ETSI EN 301 192 and ISO/IEC 13818-1 for MPEG-2 encapsulation of synchronous serial data.

1.2. (NU) GENERAL

Figure E - 1 highlights the portion of the SBS that are covered in this annex.

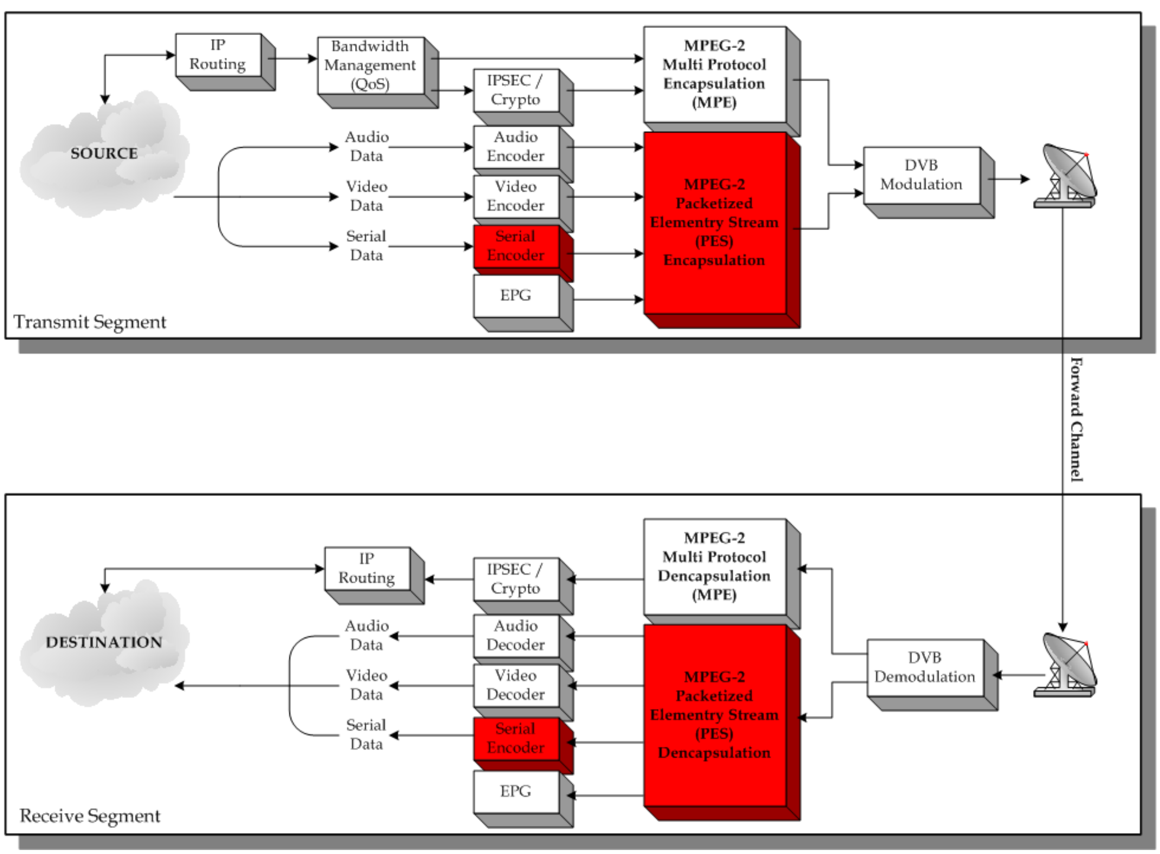

Figure E - 1 Functional Location for Serial Stream Service

2. (NU) SERIAL OVER MPEG-2 ENCAPSULATION

Serial data shall be encapsulated into MPEG-2 Transport Streams in accordance with:

- the Program Elementary Stream (PES) packets described in ETSI EN 301 192,
- the mapping of PES packets into the MPEG-2 Transport Stream as described in ISO/IEC 13818-1

This encapsulation is described below.

2.1. (NU) PACKETIZED ELEMENTARY STREAM (PES) PACKETS

In accordance with ETSI EN 301 192, serial data being broadcast in an MPEG-2 Transport Stream is carried within Program Elementary Stream (PES) packets. PES packets are mapped into the MPEG-2 Transport Stream based on the method defined in ISO/IEC 1318-1. Figure E - 2 illustrates the general process of serial stream encapsulation.

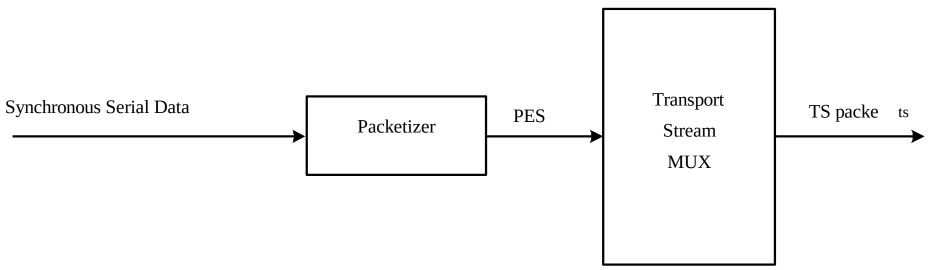

Figure E - 2 Serial Stream Encapsulation Process

Serial data is first converted into the PES_data_packet structure defined in ISO/IEC 13818-1. This enables the serial data to be inserted into PES packets. The PES_data_packet structure is shown in Figure E - 3.

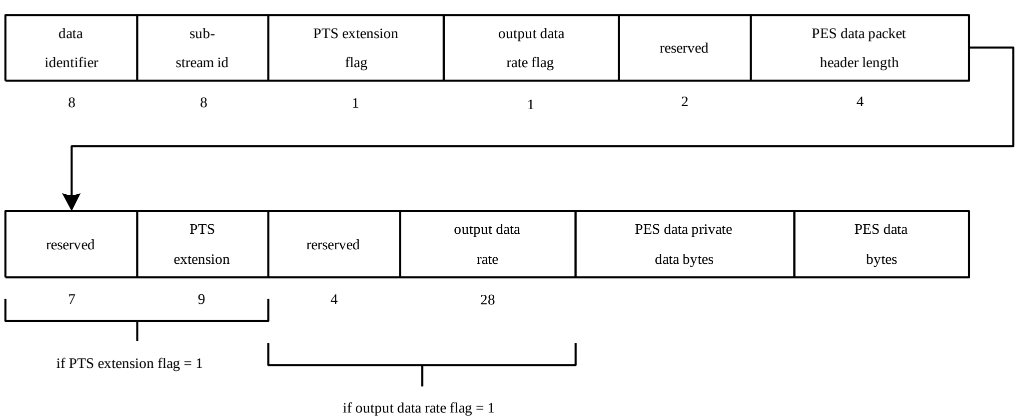

Figure E - 3 PES_data_packet structure

The data_identifier field indicates the type of data being carried in the PES packet. The data_identifier field will be set to 0x21, which indicates DVB synchronous data streams. The PES_extension _flag will be set to 1. The presence of the output_rate field will determine the value of the output_rate_flag, accordingly. The PES_data_packet _header_length indicates the length of the optional fields in the packet, including the PES _data_private_data_bytes.

PES packets consist of a header and packet data. Figure E - 4 depicts the syntax of a PES packet.

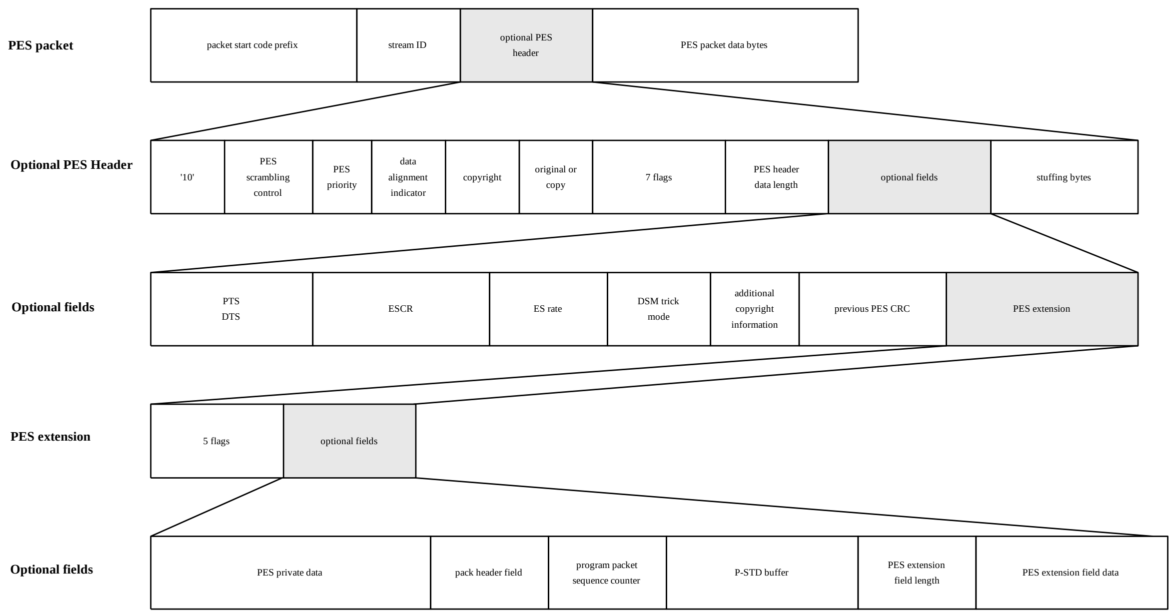

Figure E - 4 Packetized Elementary Stream (PES) Packet Syntax

The stream_id will have a value of 0xBD, which indicates a private stream. The produced PES packets will be of nonzero length. For complete semantic definitions of the PES packet see ISO/IEC 13818-1.

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ANNEX F: ANALOG VIDEO AND AUDIO ENCAPSULATION

1. (NU) INTRODUCTION

This annex describes the process used to deliver Video and Audio data via MPEG-2 Transport Streams (ref. ETSI EN 301 192, ISO/IEC 13818-1/13818-2/13818-3) as required to support DVB Modulation and Channel Coding for NATO SBS (ref. ETSI EN 300 421). Audio and Video data are characterized as being synchronized data streams. Synchronized data streams have timing requirements in the sense that the data within each stream can be played back in synchronization other kinds of data streams.

Analog video and audio supportability is not a mandatory requirement for NATO SBS. The information provided herein is required only if the option to support analog video and audio input is implemented.

1.1. (NU) AIM

This is an outline of relevant specifications provided in ETSI EN 301 192 and ISO/IEC 13818-1/13818-2/13818-3 for MPEG-2 encapsulation of audio/video data.

1.2. (NU) GENERAL

Figure F - 1 highlights the portions of the SBS that shall be covered by this annex.

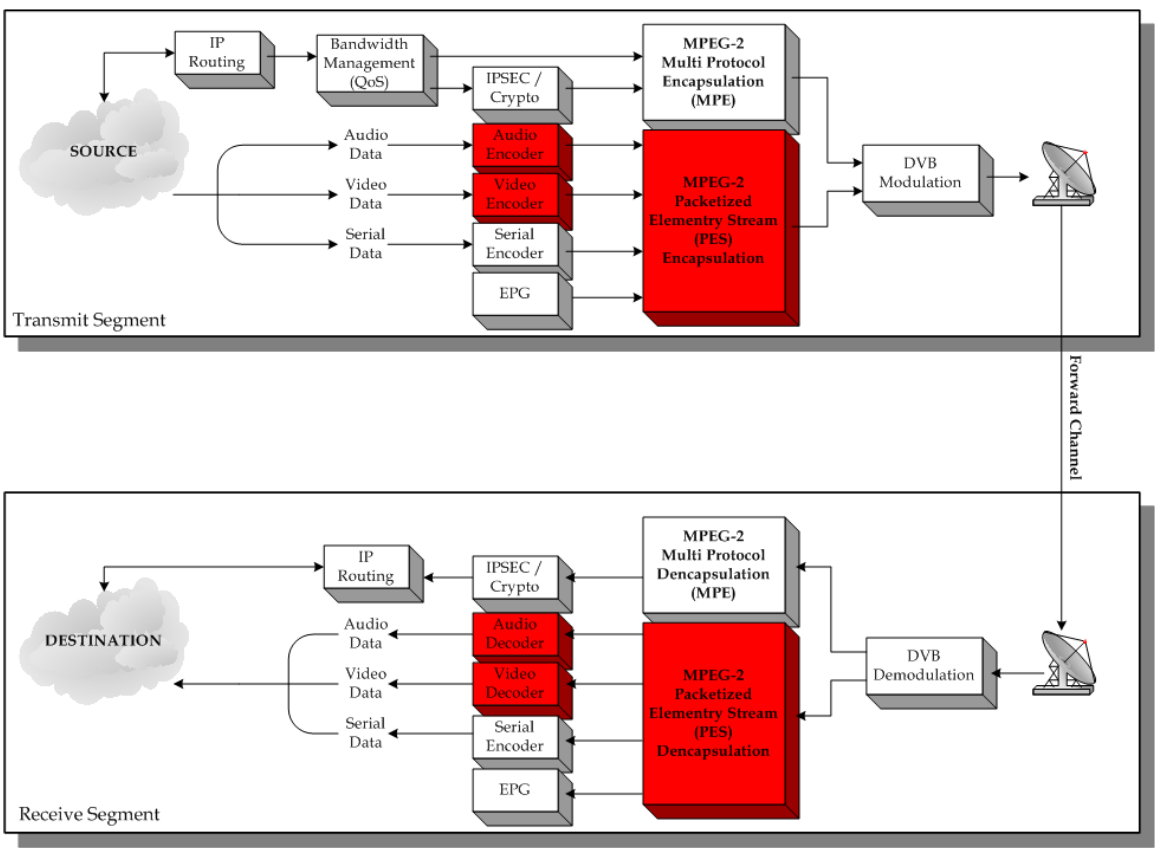


Figure F - 1 Functional Location of Audio/Video Service

2. (NU) VIDEO AND AUDIO DATA OVER MPEG-2 ENCAPSULATION

Analog Video and Audio data shall be encapsulated into MPEG-2 Transport Streams in accordance with:

- the video and audio encoding restrictions described in ISO/IEC 13818-2 and ISO/IEC 13818-3
- the Program Elementary Stream (PES) packets described in ETSI EN 301 192,
- the mapping of PES packets into the MPEG-2 Transport Stream as described in ISO/IEC 13818-1

This encapsulation is described below.

Analog Video and Audio data must be digitally encoded prior to MPEG-2 Encapsulation. Video and Audio data shall be encoded to produce compressed elementary streams, as defined in ISO/IEC 13818-2 and ISO/IEC 13818-3. These streams shall subsequently be packetized to produce Packetized Elementary Stream (PES) packets, which are then encapsulated and multiplexed to produce MPEG-2 Transport Stream packets. Figure F - 2 depicts the general process of MPEG-2 Encapsulation of Video and Audio data. The process of video and audio encoding and encapsulation is further described below.

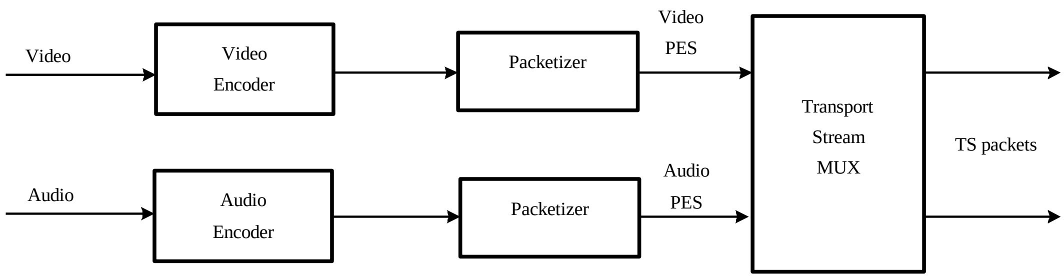


Figure F - 2 Video/Audio Encapsulation process

2.1. (NU) VIDEO ENCODING

2.1.1. (NU) Profile and Level

The encoding of video data will comply with the Main Profile Main Level restrictions, as defined in ISO/IEC 13818-2. Table F - 1 indicates the values of the profile_and_level indication for the various IRDs and bitstreams.

Table F - 1 Profile and Level Indication values for Various IRDs/Bitstreams

IRD/bitstream type	profile_and_level_indication values
25 Hz SDTV	01001000
25 Hz HDTV	01000100
30 Hz SDTV	01001000
30 Hz HDTV	01000100

2.1.2. (NU) Frame Rate

The frame rates for the various IRDs and bitstreams are indicated in Table F - 2.

Table F - 2 Frame Rates and Codes for various IRDs/Bitstreams

IRD/bitstream type	Frame Rate	Frame Rate Code
25 Hz SDTV	25 Hz	0011
25 Hz HDTV	25 Hz or 50 Hz	0011 or 0110
30 Hz SDTV	24000/1001, 24, 30000/1001, or 30Hz	0001, 0010, 0100, 0101
30 Hz HDTV	24000/1001, 24, 30000/1001, 30, 60000/1001, 60 Hz	0001, 0010, 0100, 0101, 0111, 1000

For 25 Hz HDTV IRDs, the format for 50 Hz frame rate will be progressive. The format for 25Hz frame rate will be progressive or interlaced. For 30 Hz HDTV IRDs, the format for 24000/1001, 24, 30000/1001, 30, 60000/1001, or 60 Hz frame rates will be progressive or interlaced.

2.1.3. (NU) Aspect Ratio

The source aspect ratios are defined in Table F - 3.

Table F - 3 Source Aspect Ratios for various IRDs/Bitstreams

IRD/bitstream type	Source Aspect Ratio	aspect_ratio_information
25 Hz SDTV	4:3, 16:9, 2.21:1	0010, 0011, 0100 (respectively)
25 Hz HDTV	16:9, 2.21:1	0011, 0100 (respectively)
30 Hz SDTV	4:3, 16:9, 2.21:1	0010, 0011, 0100 (respectively)
30 Hz HDTV	16:9, 2.21:1	0011, 0100 (respectively)

In the case of the 25 Hz SDTV and 30 Hz SDTV, if the source aspect ratio is 16:9 or 2.21:1, it is recommended that pan vectors for a 4:3 window be included in the bitstream. If pan vectors are included, the sequence_display_extension field will also be present in the bitstream and the aspect_ratio_information field will be set to "0010". The display_vertical_size will be equal to the vertical_size. The diplay_horizontal_size will be equal to:

display_horizontal_size = 4/3 * horizontal_size/source aspect ratio

2.1.4. (NU) Luminance Resolution

For 25 Hz SDTV bitstreams, the encoded picture will have a full-screen luminance resolution of 720 x 576, 544 x 576, 480 x 576, 352 x 576, or 532 x 288. For 25 Hz and 30 Hz HDTV IRDs, the encoded picture will not have more than 1152 lines per frame, 1920 luminance samples per line, or 62,668,800 luminance samples per second. It is recommended that 25 Hz bitstreams, have a luminance resolution of 1,082 lines per frame and 1,920 luminance samples per line, with an associated frame rate of 25 Hz. For 30 Hz bitstreams, it is recommended that the luminance resolution be 1080 lines per frame and 1,920 luminance samples per line be utilized for frame rates of 30000/1001 Hz with interlaced fields per frame. For 25 Hz Encoded pictures using 30 Hz SDTV IRDs will have a full-screen luminance resolution of 720 x 480, 640 x480, 544 x 480, 480 x 480, 352 x 480, or 352 x 240.

2.1.5. (NU) Chromaticity Parameters

For 25 Hz SDTV bitstreams, if the sequence_display_extension() is not present in the bitstream or the colour_description is zero, the chromaticity is implicitly defined to be corresponding to the colour_primaries with a value of 5, the transfer characteristics will be implicitly defined to be corresponding to the transfer_characteristics with a value of 5, and the matrix coefficients will also be implicitly defined to have a value corresponding to the matrix_coefficients with a value of 5.

For 25 Hz and 30 Hz HDTV, the colour_primaries, transfer_characteristics, and matrix_coefficients will have a value of 1, as indicated in ETSI TR 101 154. Within 30 Hz SDTV bit streams, if the sequence_display_extension() is not present in the bit stream or colour_description is zero, the chromaticity shall be implicitly defined to be that corresponding to colour_primaries having the value 6, the transfer characteristics shall be implicitly defined to be those corresponding to transfer_characterstics having the value 6 and the matrix coefficients shall be implicitly defined to be those corresponding matrix_coefficients having the value 6. This set of parameter values signals compliance with SMPTE 170M

2.1.6. (NU) Chrominance

Chroma_420_type will be set to zero if the fields are down sampled independently or one if the two fields have been combined into a single frame prior to down sampling. It is recommended that the fields be down sampled independently for all bitstream types (25 Hz SDTV, 25 HDTV, 30 SDTV, 30 HDTV)

2.1.7. (NU) Video Sequence Header

The video sequence header will be encoded once every 500 milliseconds for all bitstream types. If non-default quantizer matrices are utilized, the appropriate intra_quantizer_matrix and/or non_intra_quantizer_matrix should be included in every sequence header.

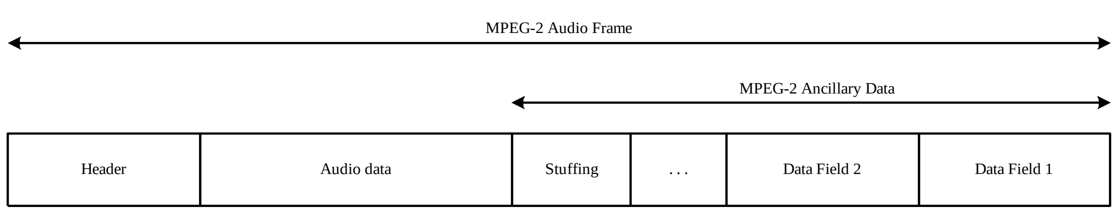
2.2. (NU) AUDIO ENCODING

Audio data is encoded in compliance with ISO/IEC 13818-3. Audio will be encoded using a multi-channel audio mode. The bitstream will use Layer II coding by setting the value of the layer field to be "10". The bitrate_index will have a value between "0001" and "1110" inclusive, which correspond to bit rates of 32, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320, and 384 Kbits/sec. For encoded bitstreams with larger bit rates, an extension bitstream will be used, with the bit rate of the extension ranging from 0 to 682 Kbit/sec.

Primary sound services will have an audio sampling rate of 32 kHz, 44.1 kHz, or 48 kHz. Secondary sound services will have an audio sampling rate of 16kHz, 22.05 kHz, 24 kHz, 32 kHz, 44.1 kHz, or 48 kHz.

The emphasis field of the encoded bitstream will have a value of "00". The crc_check will be included in the bitstream. The bitstream will not use emphasis, mc_prediction, and multilingual channels. The value of those fields will be set to zero.

Encoded bitstreams may contain an ancillary data field. The insertion of data fields begins at the end of the MPEG-2 audio frame. Figure F - 3 depicts the use of an ancillary data field as shown in ETSI TR 101 154.



2.3. (NU) PACKETIZED ELEMENTARY STREAM (PES) PACKETS

Figure F - 3 Use of ISO/IEC 13818-3 Ancillary data field

Following the encoding of video and audio data, the bitstreams are multiplexed to produce compressed elementary streams. Elementary Stream data is carried in Packetized Elementary Stream (PES) packets. PES packets consist of a header and packet data. Figure F - 4 depicts the syntax of a PES packet.

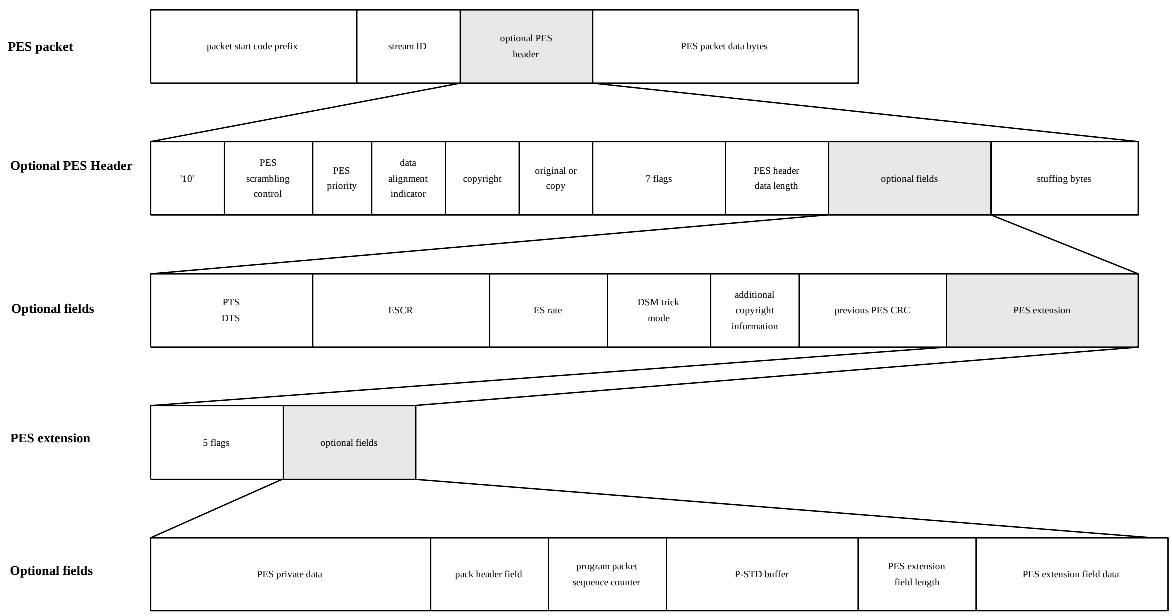


Figure F - 4 Packetized Elementary Stream packet

Elementary streams will be indicated by a stream_id, as shown in Table F - 4.

Table F - 4 Stream_id values for audio and video data

Data type	stream_id	stream coding
Video	1110 xxxx	ISO/IEC 13818-2 video stream number xxxx
Audio	110x xxxx	ISO/IEC 13818-3 audio stream number x xxxx

The stream_type will be set to 0x02 and 0x04 for video and audio data respectively.

2.4. (NU) IRD DECODING

The Integrated Receiver-Decoder (IRD) will decode the bitstreams when they are received at the Receive Suite. The following sections outline the guidelines for IRDs for both Standard Definition Television (SDTV) and High Definition Television (HDTV). All baseline IRDs must be able to demultiplex the MPEG-2 transport stream.

2.4.1. (NU) Transport Stream System Target Decoder (T-STD)

The IRD must be capable of managing clock frequencies ranging from:

27,000,000 - 810 <= system_clock_frequency <= 27,000,000 + 800, where
the rate of change of system_clock_frequency with time <= 75 x 10-3 Hz/s

2.4.2. (NU) Transport Packet Header

When the transport_error_indicator flag is set to one, the IRD must perform or call an error recovery mechanism. The transport_priority field is ignored by the IRD. The IRD will act accordingly to the scrambling of the bits. The transport_scrambling_control field will indicate the scrambling used during encoding.

2.4.3. (NU) Adaptation Field

The elementary_stream_priority_indicator field will be ignored by the IRD. The IRD must have the capability of operating with PCRs arriving at intervals not exceeding 100 milliseconds.

2.4.4. (NU) PES Packet

The IRD will act accordingly to the scrambling of the PES packets, as indicated by the PES_scrambling_control field. The PES_priority field will be ignored by the IRD. The copyright and original_or_copy fields will not be altered by the IRD. The ESCR and ES rate fields are not decoded by the IRD.

2.4.5. (NU) Program and Elementary Stream Descriptors

The IRD will use the video_stream_descriptor and the audio_stream_descriptor fields to determine decoding procedures. If the target_background_grid_descriptor is not present, the default grid of 720 x 576 pixels or 720 x 480 pixels will be assumed for a 25 Hz bitstream and a 30 Hz bitstreams respectively. The IRD will interpret the video_window_descriptor, the CA_descriptor, and the ISO_639_Language_descriptor; All other descriptors can be ignored by the IRD.

ANNEX G: DATA ENCRYPTION

1. (NU) INTRODUCTION

Data encryption shall be implemented as required to support the security enclave that the Satellite Broadcast Service is interfaced with. NATO SBS systems must support Type 1 IP encryption devices. As an option, bulk encryption may be used.

1.1. (NU) AIM

This annex describes data encryption methods for NATO SBS systems.

1.2. (NU) GENERAL

Figure G - 1 highlights the portion of the SBS that shall be covered in this annex.

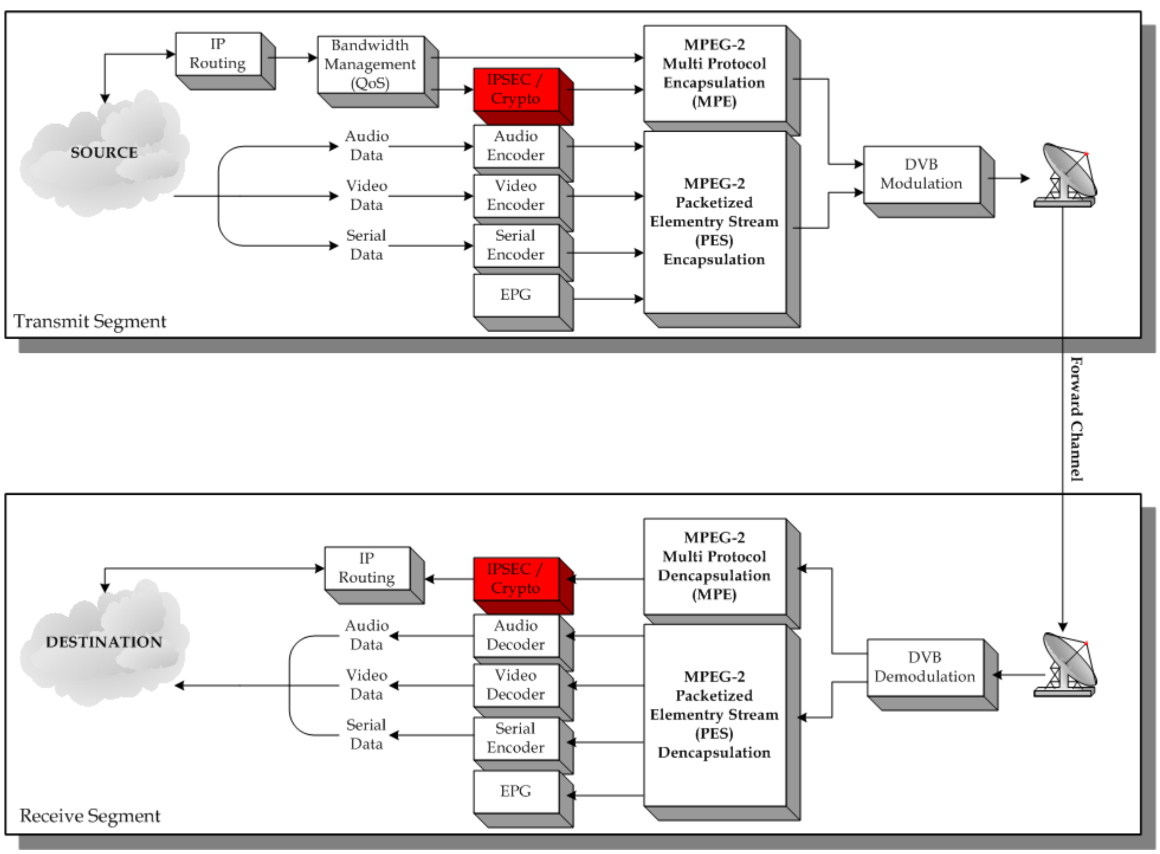


Figure G - 1 Functional Location for IP Data Encryption

2. (NU) IP ENCRYPTION

NATO SBS systems shall use HAIPIS-compliant Type-1 IP-encryption devices. NATO SBS systems shall support the use of the Encapsulating Security Payload (ESP) operating in Tunnel Mode as defined in RFC 4301 with Tunnel Mode Security Association (SA) as defined in HAIPIS. This format is described below.

2.1. (NU) SECURITY ASSOCIATIONS

A SA is a connection that allows security services to be implemented on the traffic being carried. There are two types of SAs: Transport Mode and Tunnel Mode. The tunnel mode SA is applied to an IP tunnel. There is an outer IP header that indicates the IPsec destination, along with an inner IP header that indicates the actual destination of the packet. For the Encapsulating Security Payload (ESP) protocol, protection only spans the tunneled packet and does not include the outer header.

2.2. (NU) ENCAPSULATING SECURITY PAYLOAD

The ESP header is inserted after the IP header and prior to the encrypted IP packet. ESP allows for confidentiality, data origin authentication, integrity, and anti-replay service.

2.2.1. (NU) ESP Packet Format

Figure G - 2 depicts the general syntax for the ESP packet.

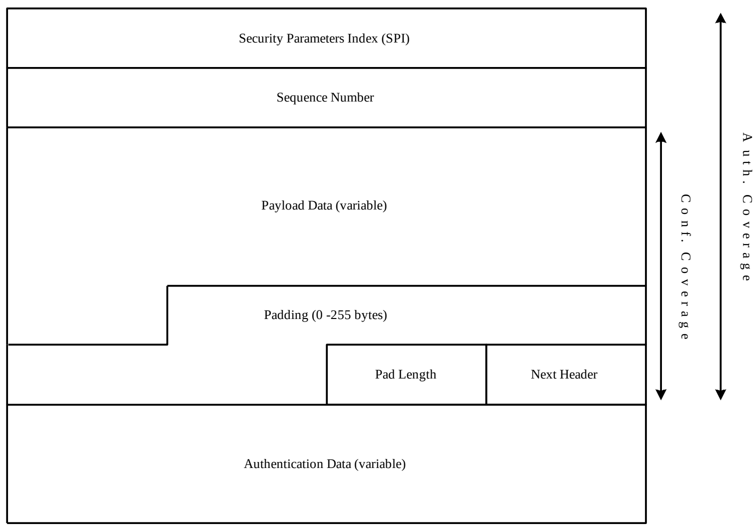


Figure G - 2 Encapsulating Security Payload Packets

The IP header will have the value of 50 in its Protocol data field to indicate the presence of the ESP header. The Security Parameters Index (SPI) is a 32-bit field that uniquely identifies the SA, when used in conjunction with the destination IP address and security protocol, which in this case is ESP. SPI values of 1 through 255 are reserved for the use of the Internet Assigned Number Authority (IANA). The SPI value of zero is reserved for local user and will not be sent.

The sequence number is an unsigned 32-bit field that is used as a counter. It is mandatory for the sender to transmit this field. However, the receiver does not need to process it if the anti-replay service is disabled.

The padding field is used for encryption. Its purpose is to fill the Payload to the size required by the encryption algorithm. The pad length field ranges from 0 – 255 bits and indicates the number of pad bytes preceding it. The next header is an 8-bit field that describes the content of the payload.

The authentication data is a variable length field containing an Integrity Check Value (ICV) computed for the ESP packet, not including the authentication data.

Figure G - 3 depicts the encapsulation of the IP packets by applying the ESP protocol.

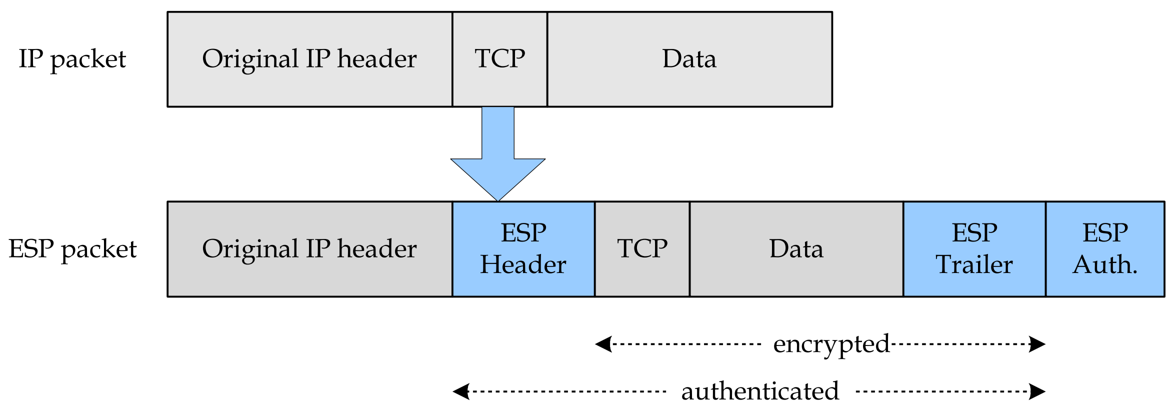


Figure G - 3 ESP Encapsulation in Transport Mode

In tunnel mode, ESP protects the entire IP packet. In this scenario, there will be an inner and outer IP header. The inner header will carry the actual destination address. The outer header may contain addresses of the security gateways. Figure G - 4 depicts the ESP encapsulation of IP packets in tunnel mode.

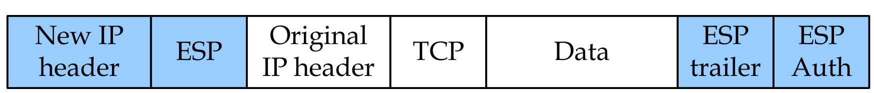


Figure G - 4 ESP Encapsulation in Tunnel Mode

2.2.2. (NU) Packet Encryption

During packet encryption, the entire original IP packet is encapsulated and the necessary padding is added. Once this is complete, the packet is encrypted using the key, encryption algorithm, and algorithm mode.

2.2.3. (NU) Inbound Packet Processing

If the packets were fragmented, re-assembly will occur prior to ESP processing. Using the destination IP address, ESP, and SPI, the receiver can determine the appropriate SA. If no valid SA exists, the packet must be discarded, and the event will be logged.

If anti-replay is enabled, which is the default, the receive-packet counter will be initialized to zero. For each packet that arrives, the receiver will verify that the packet has a unique Sequence Number. If any duplicate packets arrive, the packets will be rejected, using the sliding window. A window size of 64 is preferred, but a minimum size of 32 is necessary.

If authentication has been selected, the receiver will compute the Integrity Check Value (ICV) over the ESP packet, not including the Authentication data. If the test fails, the datagram is discarded and the event should be logged.

The receiver must then decrypt the ESP Payload data, padding, pad length, and next header using the key, encryption algorithm, algorithm mode, and cryptographic synchronization data, which is indicated by the SA. Following the decryption, the receiver must process and remove any padding that was added during encryption. Once these steps have been complete, the original IP datagram is reconstructed.

3. (NU) CONDITIONAL ACCESS (CAS)

A Conditional Access System (CAS) may be used to alter the characteristics of video, audio, and IP data being carried within the MPEG-2 Transport Stream in order to prevent unauthorized reception of the information. The implementation of CAS must be done in accordance with ISO/IEC 13818-1. This section describes the standard mechanisms to be used by service providers that will be transporting and decoding conditional access data via NATO SBS.

3.1. (NU) CONDITIONAL_ACCESS_SECTION FORMAT

Whenever one or more elementary streams within an MPEG-2 Transport Stream are scrambled, Transport Stream packets with a PID value of 0x0001 shall be transmitted with a complete Conditional Access Table and CA_descriptors in the form of a conditional_access_section as defined in ISO/IEC 13818-1. The Conditional Access Table (CAT) provides the association between one or more EMM streams and their related parameters. The CAT can consist of one or many of the conditional_access_section syntax shown in Figure G - 5.

Table ID	Section Syntax Indicator	‘0’		Section Length	Version Number	Current Next Indicator	Section Number	Last Section Number	N-Loop Descriptors	CRC
{8}	{1}	{1}	{2}	{12}	{5}	{1}	{8}	{8}		{32}

Figure G - 5 Conditional_Access_Section Syntax

The conditional access descriptor is used to specify both system-wide conditional access management information such as EMMs and elementary stream-specific information such as ECMs.

3.1.1. (NU) Entitlement Control Message (ECM)

ECMs contain private conditional access information, which specify control words and possibly other, typically stream-specific, scrambling and/or control parameters.

3.1.2. (NU) Entitlement Management Message (EMM)

EMMs contain private conditional access information, which specify the authorization or the services of specific decoders. They may be addressed to a single decoder or groups of decoders.

4. (NU) BULK ENCRYPTION

Encryption of classified information will use NATO approved Type-1 bulk encryption devices.

Encryption of unclassified information encryption will be optional. If encryption is implemented Type-2 AES256 is recommended.

Encryption of the control signaling between the hub and remote terminals known as Transmission Security (TRANSEC) is optional. TRANSEC must be able to be disabled for SBSs that have TRANSEC implemented.

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ANNEX H: DVB-S MODULATION

1. (NU) INTRODUCTION

This annex describes the process of DVB modulation and channel coding, as defined in ETSI EN 300 421. This modulation method may be used exclusively or in conjunction with the second generation format described in Appendix J. Data to be broadcast using the SBS will be MPEG-2 encapsulated prior to modulation.

1.1. (NU) AIM

This annex defines the details of the DVB Modulation process (DVB-S).

1.2. (NU) GENERAL

Figure H - 1 highlights the portion of the SBS that are described in this annex.

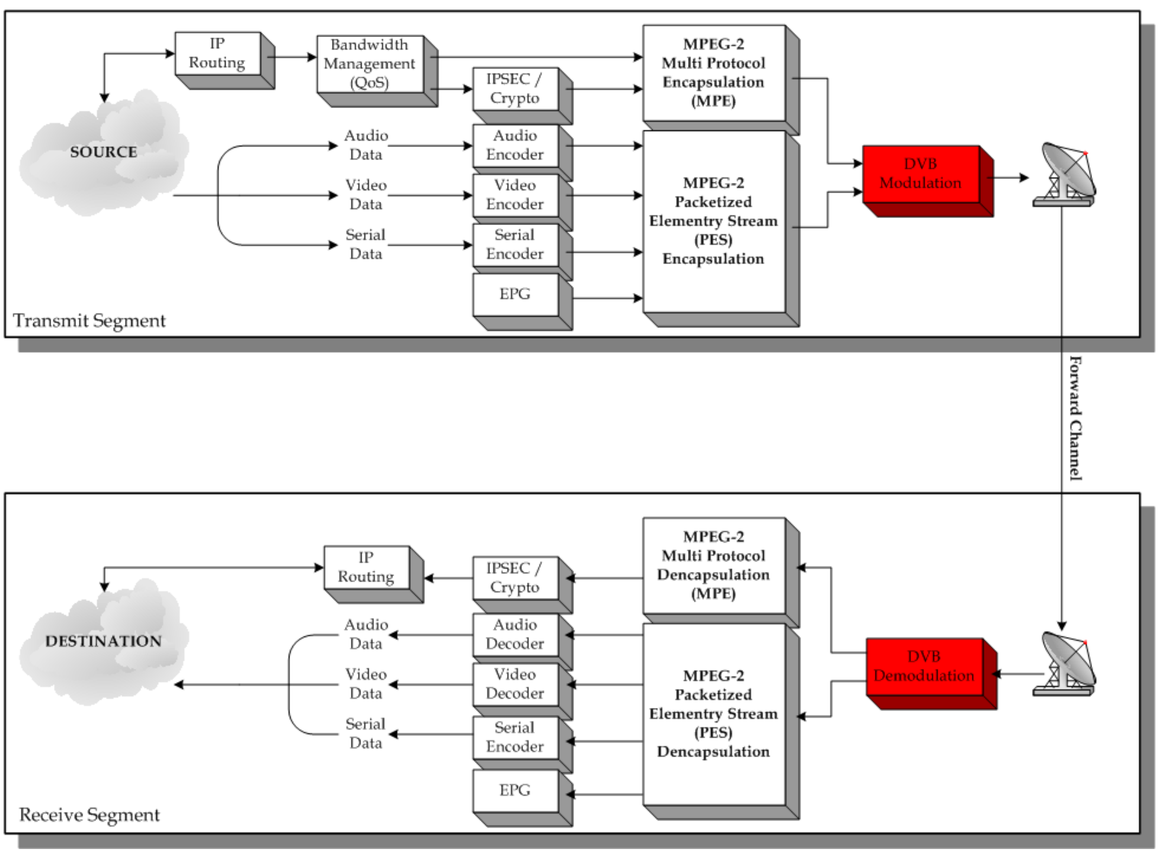


Figure H - 1 DVB Modulation Conceptual Overview

2. (NU) DVB MODULATION

Forward channel data sent over the SBS utilizing the waveform described in this annex will be transmitted according to the Digital Video Broadcast over Satellite Standard (DVB-S). DVB-S relies on MPEG-2 data packets, thereby establishing that the final encapsulation layer for all service types will be an MPEG-2 encapsulation. Once the data has been formatted to MPEG-2, it is coded and modulated according to the DVB-S commercial standards for broadcast. This process includes the following steps: Multiplexing, Energy Dispersal, Outer Coding, Interleaving, Inner Coding, Baseband Shaping, and Modulation, as shown in Figure H - 2.

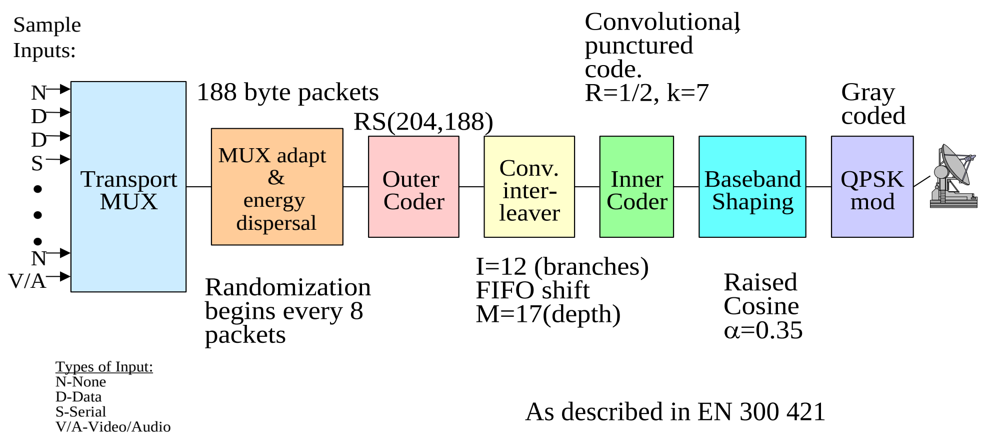


Figure H - 2 DVB-S Channel Coding and Modulation Process

2.1. (NU) TRANSPORT MULTIPLEXING

The input stream shall be organized in fixed length packets following the MPEG-2 transport multiplexer (see ISO/IEC DIS 13818-1 and EN 300 421). The input stream is described below.

The transport multiplexer will multiplex together 1 to n MPEG-2 service Transport Streams. The number n is dependent on the system. The resulting MPEG-2 data stream out of the transport multiplexer will be a Transport Stream carrying fixed 188 byte sized packets. Each packet begins with a sync-word byte of 47_{HEX} (01000111). The processing order at the transmitting side always starts from the MSB (i.e "0") of the sync word-byte (i.e. 01000111). Figure H - 3 shows the MPEG-2 Transport Packet.

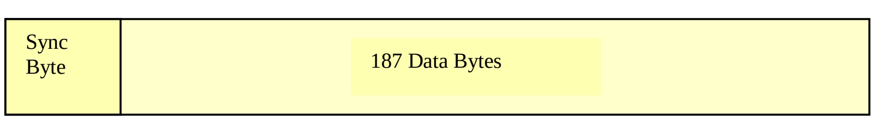


Figure H - 3 MPEG-2 Transport Packet

2.2. (NU) TRANSPORT MULTIPLEX ADAPTATION AND RANDOMIZATION

Transport Multiplex Adaptation and Randomization of the stream of MPEG-2 packets shall be performed in accordance with Section 4.4.1 of EN 300 421. This process is described below.

To coordinate the randomization/de-randomization process, the MPEG-2 DVB stream goes through a process called transport multiplex adaptation. This involves bit-wise inverting the first sync byte of every eight MPEG-2 transport packets. The byte would then become $B8_{HEX}$ (inverted from 47_{HEX}). Transport multiplex adaptation provides a means for initializing the randomizer/de-randomizer. A $B8_{HEX}$ byte triggers the loading of "100101010000000" into a Pseudo Random Binary Sequence (PRBS) generator. The timing of the loading sequence is such that the MSB of the MPEG-2 packet following the inverted sync byte is "ex-ored" with the first output bit of the generator. During the processing of the sync word bytes of the following seven transport packets, the PRBS generator will remain active. However, the randomizer will be disabled in order to preserve synchronization purposes of these bytes in the system.

Thus, the period of the PRBS generator will be 1503 Bytes, beginning at the end of the inverted sync byte on packet 1, and ending at the end of packet 8. Figure H - 4 shows a sample of a randomized DVB stream.

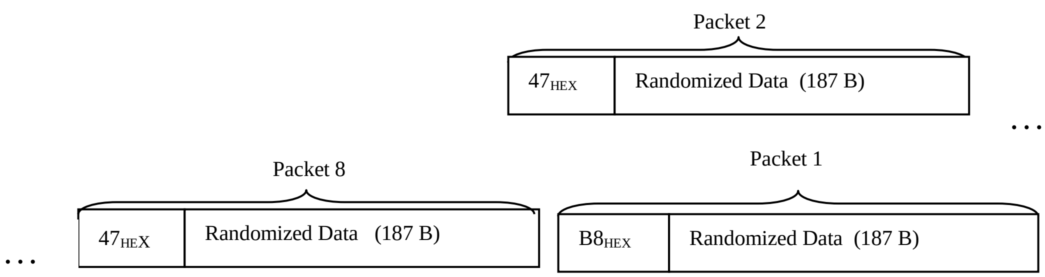


Figure H - 4 Randomized DVB Stream

2.3. (NU) OUTER CODING

Outer coding shall be performed in accordance with Section 4.4.2 of EN 300 421. This coding is described below.

Outer coding consists of Reed Solomon RS(204,188,t=8) shortened code applied to each randomized MPEG-2 transport packet, including inverted and non-inverted sync bytes. The shortened Reed Solomon code is generated by adding 51 null bytes to the beginning of the 188 byte MPEG transport packet at the input of a standard k=8, RS(255,239) encoder. When the encoding is finished, the null bytes are discarded leaving the RS(204,188, T=8) error protected packet.

The code generator polynomial is $g(\chi) = (\chi + \lambda^0)(\chi + \lambda^1)(\chi + \lambda^2)(\chi + \lambda^3) \ \ldots (\chi + \lambda^{15})$, where $\lambda = 02_{HEX}$, and the filed generator polynomial is $p(\chi) = \chi^8 + \chi^4 + \chi^3 + \chi^2 + \chi^1$.
Figure H - 5 shows an example of a DVB error protected packet.

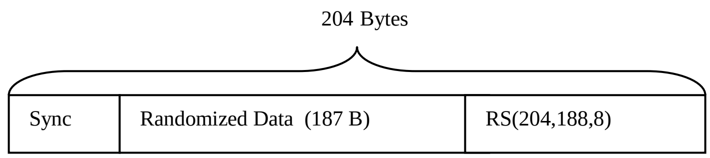


Figure H - 5 Reed Solomon Error Protected Packet

2.4. (NU) INTERLEAVING

Interleaving shall be performed in accordance with Section 4.4.2 of EN 300 421. The interleaving process is described below.

Convolutional interleaving with an interleaving depth of I=12 bytes is applied to the RS coded packets in accordance with the EN 300 421 DVB modulation and channel coding description. The interleaver may be composed of 12 branches, each operating in a First-In, First-Out (FIFO) shift. Each branch will have depth of (M j) cells (M=17=N/I, N=204=error protected frame length, I=12=interleaving depth, j = branch index). The cells each contain one byte.

An input switch cyclically connects the input byte-stream to the interleaver branches. The input and output switches of the interleaver must be synchronized.

The sync bytes (inverted and non-inverted) are always routed through the “0” branch for synchronization purposes. The resulting interleaved frame is composed of overlapping error-protected packets defined by the sync bytes. Figure H- 6 displays a sample packet.

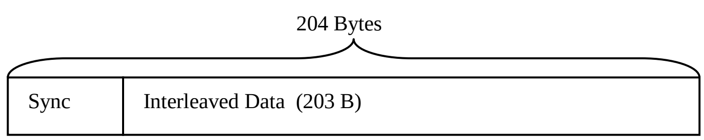


Figure H- 6 Interleaved Packets

2.5. (NU) INNER CODING

Inner Coding shall be performed in accordance with Section 4.4.3 of EN 300 421. The inner coding process is described below.

A range of punctured convolutional codes are supported, based on a rate one half convolutional code with constraint length K = 7. This flexibility allows for adjustment for an appropriate level of error correction. Rates 1/2, 2/3, 3/4, 5/6, and 7/8 are supported as defined in Figure H - 7.

Original Code			Code Rates									
			1/2		2/3		3/4		5/6		7/8	
k	$G_1(X)$	$G_2(Y)$	P	d_{free}	P	d_{free}	P	d_{free}	P	d_{free}	P	d_{free}
7	171oct	133oct	X: 1 Y: 1 $I = X_1$ $Q = Y_1$	10	X: 1 0 Y: 1 1 I = X1 Y2 Y3 Q = Y1 X3 Y4	6	X: 1 0 1 Y: 1 1 0 I = X1 Y2 Q = Y1 X3	5	X: 1 0 1 0 1 Y: 1 1 0 1 0 I = X1 Y2 Y4 Q = Y1 X3 X5	4	X: 1 0 0 0 1 0 1 Y: 1 1 1 1 0 1 0 I = X1 Y2 Y4 Y6 Q = Y1 Y3 X5 X7	3
NOTE: 1 = transmitted bit 0 = non transmitted bit												

Figure H - 7 Punctured Code Definition

2.6. (NU) Baseband Shaping

Baseband shaping shall be performed in accordance with Section 4.5 of EN 300 421. The shaping process is described below.

Prior to modulation, the I and Q signals at a symbol rate Rs are passed through a baseband square root raised cosine filter with a theoretical response to a sequence of Dirac delta functions as defined by

$$H(f) = \begin{cases} 1 & |f| < f_N(1-\alpha) \\ \left\{ \dfrac{1}{2} + \dfrac{1}{2} \sin \dfrac{\pi}{2f_N} \left[\dfrac{f_N - |f|}{\alpha} \right] \right\}^{1/2} & f_N(1-\alpha) \le |f| \le f_N(1+\alpha) \\ 0 & |f| > f_N(1+\alpha) \end{cases}$$

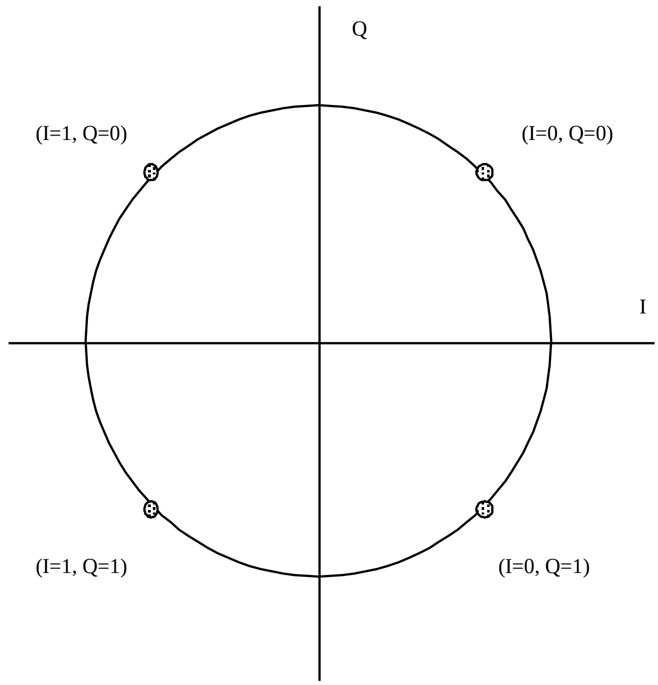
where $f_N = R_s/2$, the Nyquist frequency, and $\alpha = 0.35$ is the roll-off factor. The filter response is based on the assumption of ideal Dirac delta functions input at rate R_s.

2.7. (NU) Modulation

Modulation shall be performed in accordance with Section 4.5 of EN 300 421. This modulation is described below.

The final step of DVB formatting for broadcast is modulation. DVB modulation employs Gray-coded, non-differential, Quadrature Phase Shift Keying (QPSK). Bit mapping in the signal space is given in Figure H - 8.

Figure H - 8 QPSK Constellations



2.8. (NU) DEMODULATION

Once the data has been received at the Receive Terminal, DVB demodulation will occur to retrieve the data. Figure H - 9 depicts the overall process of demodulation.

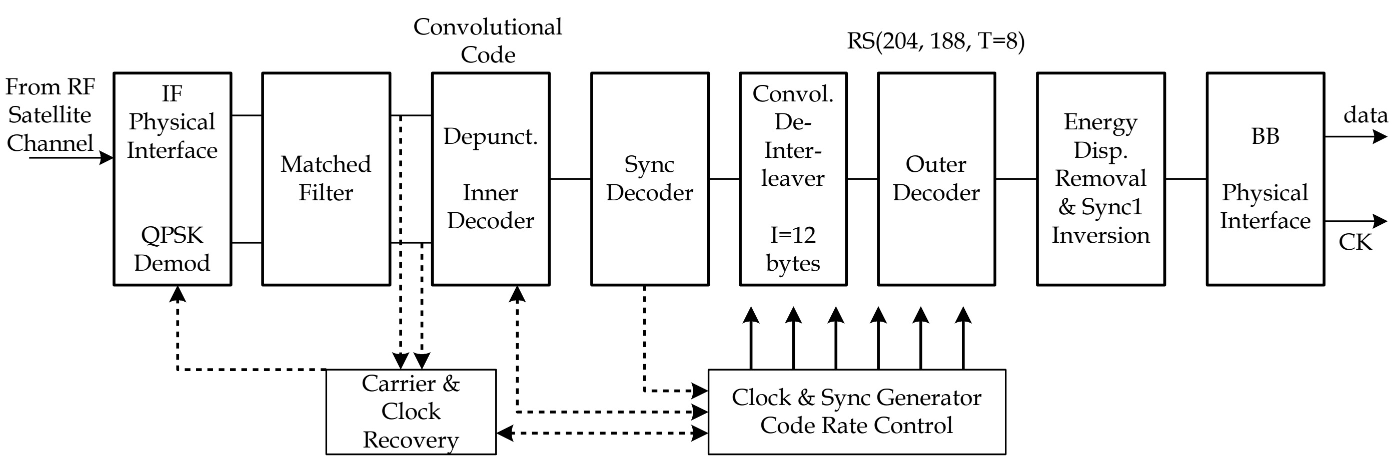


Figure H - 9 DVB Demodulation Process, as specified in ETSI EN 300 421

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ANNEX I: DVB-RCS (RETURN CHANNEL SATELLITE)

1. (NU) INTRODUCTION

This annex describes the implementation of an optional DVB return channel via satellite (DVB-RCS), as defined in ETSI EN 301 790 V1.4.1 (2005-04).

1.1. (NU) AIM

This annex provides a general overview of the DVB-RCS process as described in EN 301 790. Specific emphasis is placed on implementation details required and recommended for SBS systems to ensure interoperability. Features and operations described herein but not defined in EN 301 790 are provided to facilitate interoperability or performance and are appropriately referenced.

1.2. (NU) GENERAL

DVB-RCS fully exploits the use of satellite systems with added return channels that allow Return Channel Satellite Terminals (RCST) to engage in a 2-way communication with the hub. Interactive applications such as VoIP, video streaming, dynamic IP routing, Virtual Private Networking (VPN), and TCP sessions that enable FTP and Web services can now be deployed to a wide range of users that previously and typically communicated back over a terrestrial link with considerable delay much larger than satellite round trip delay.

DVB-RCS follows standard ETSI EN 301 790 which implements the underlying concept of Multiple-Frequency Time Division Multiple Access (MF TDMA). Time and frequencies are mapped and scheduled for the return traffic such that all bandwidth channels are efficiently utilized by remote users.

This appendix includes features that are not specified in EN 301 790, but provide functionality necessary for critical SBS applications or improve system performance or efficiency. Guidance provided in the SatLabs System Recommendations has been incorporated in this annex to clarify ambiguity in EN 301 790 and ensure interoperability of basic system functions. Basic functionality as described in this annex includes:

- Ability to log on to the DVB-RCS network
- Maintaining synchronization with the network
- Requesting return channel capacity and utilizing allocated channel resources
- Transport of IP datagrams to and from the gateway

Features that are described to enhance performance or provide additional services are listed as optional, and are not mandatory in a SBS system.

2. (NU) DVB-RCS SYSTEM

The DVB-RCS network consists of broadcast channels and interactive channels.

Broadcast channels are unidirectional forwarding channels which provide point-to-multipoint services delivering content from the service provider to the users behind RCSTs. The forward link is implemented according to DVB-S or DVB-S2 specifications as defined in EN 302 307, EN 300 421, TR 101 202, ETS 300 802, EN 300 468, EN 301 192, and TR 101 154. The choice of broadcast standard to be implemented on the forward channel is left to the system integrator. Specifics for incorporating the interactive channel with each forward channel standard are provided in EN 301 790 and this annex.

The interactive channel is a bi-directional interaction channel established between the content provider and the RCSTs to deliver information from the provider to the users on the forward path, and to deliver requests and traffic data from the RCSTs to the content provider on the return path. The interactive channel implements the DVB-RCS standard. Two components make up the interactive channel:

Return Interaction Path – connection from the user to the service provider, which is utilized by the user to make requests, answer question, or transfer data to the service provider.

Forward Interaction Path – connection from the service provider to the user, which is utilized to provide information from the service provider to the user(s) and any other required communication for the interactive service provision. It may be embedded into the Broadcast Channel. This channel is not required in simple one-way broadcast implementations, which make use of the Broadcast Channel for the carriage of data to the user.

2.1. DVB-RCS ARCHITECTURE

DVB-RCS broadcast and interactive channel equipment may be co-located or geographically separated. The ubiquitous nature of the signals provides wide latitude for the system integrator for physical and logical location of systems as permissible by satellite footprints. The following sections provide examples of each architecture.

2.1.1. Co-Located Gateway and Feeder Architecture

Figure I - 1 conceptually illustrates a system with a single broadcast channel and single traffic gateway (which processes signaling and traffic for the interactive channel). The NCC may also be co-located, or may remotely manage the system. The feeder channel is generated by the broadcast network adaptor according to the DVB-S standard described in Annex H or the DVB-S2 standard described in Annex J whereas the interactive channel is generated by the interactive network adaptor according to the DVB-RCS standard described in this annex.

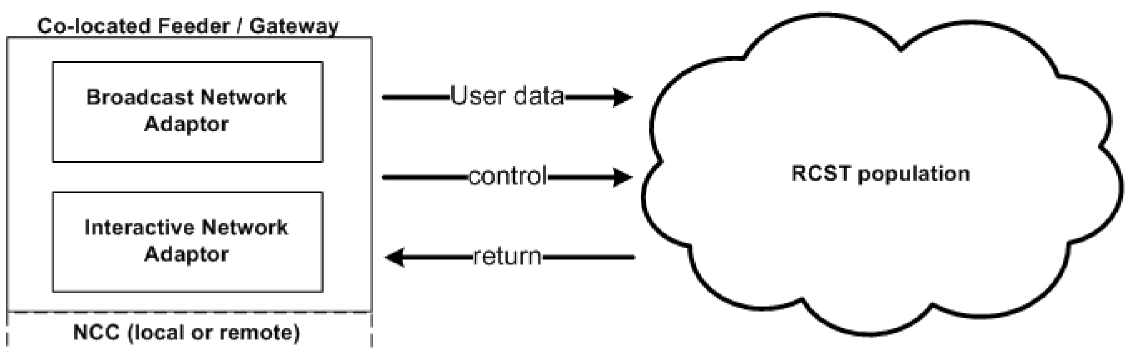

Figure I - 1 Co-Located Feeder/Gateway Architecture

2.1.2. Multiple Feeder Architecture

Figure I - 2 conceptually illustrates a network with feeder stations. The primary feeder shall broadcast control and timing signals (including timing for the associated gateway station), and may provide data traffic. Secondary feeders provide user data and may include primary feeders from other networks or theater injection points. Terminals that are equipped with two or more receivers may switch between feeds without losing network synchronization. One receiver shall be continuously tuned to the primary feeder station for control and synchronization signaling as well as monitoring, accounting, and billing. The other receivers may be tuned to any secondary feeds, limited only by the terminal’s ability to receive separate signals. This architecture is suitable for SBS scenarios where more than one injection point exists.

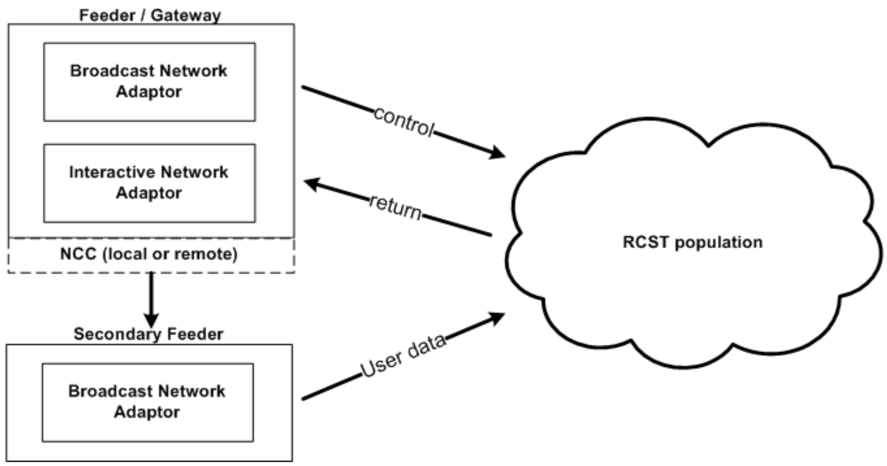

Figure I - 2 Multiple Feeder/Gateway Architecture

2.2. DVB-RCS SIGNAL PROCESSING

Figure I - 3 depicts the generic digital signal processing that occurs at the RCST transmitter side. The RCST accepts data and Medium Access Control (MAC) information for transport to the gateway and prepares it for satellite transmission. The steps in this process include burst formatting, energy dispersal, channel coding for error detection and correction, and modulation. A synchronization element in the RCST ensures proper network timing of all RCSTs with the gateway. The following sections detail the steps of this process.

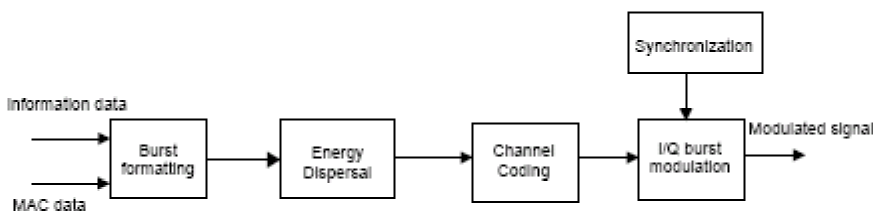

Figure I - 3 Return Channel Digital Signal Processing Flow

2.2.1. (NU) Timing and Synchronization

The DVB-RCS synchronization scheme is based on a Network Clock Reference (NCR) provided by the gateway in the forward link signal. The NCR is derived from the Network Control Center (NCC) clock and distributed within a specific logical channel identified by a program identifier (PID) in the forward link signaling stream. This NCR information provides a 27 MHz reference of the NCC clock to the RCSTs. The point of reference for the NCR information is dependant upon the transmission mode (Constant Coding and Modulation [CCM] or Adaptive Coding and Modulation [ACM]) employed on the forward link.

2.2.1.1. CCM System Timing

For systems that employ DVB-S or DVB-S2 CCM on the forward link, NCR distribution follows the Program Clock Reference (PCR) distribution mechanism as defined in ISO/IEC 13818-1.

2.2.1.2. ACM System Timing

For systems that employ DVB-S2 with ACM, a reference time axis is constructed using the DVB-S2 frame structure. In this case, the RCST associates the received NCR value with a system dependant reference symbol. The reference symbol shall be the first symbol of the Start-of-Frame (SOF) field of the N^{th} DVB-S2 frame for an NCR field of which the most significant bit is contained in the $(N+2)^{nd}$ DVB-S2 frame. Figure I - 4 illustrates the NCR packet and its corresponding reference point.

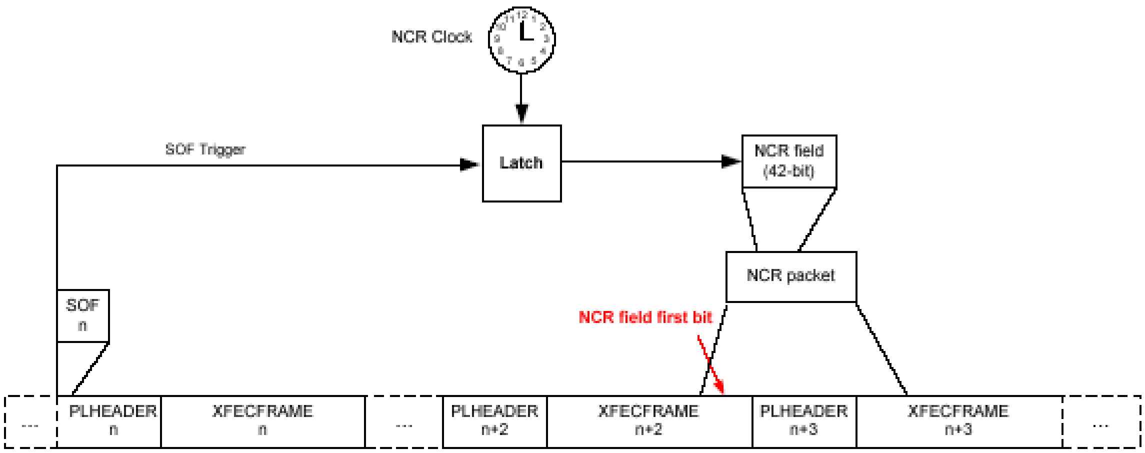

Figure I - 4 NCR Timing Reference for DVB-S2 Forward Link

2.2.2. Burst Formatting

The DVB-RCS network uses four types of bursts:

- Traffic bursts (TRF)
- Acquisition bursts (ACQ)
- Synchronization bursts (SYNC)
- Common Signaling Channel bursts (CSC)

Burst type is determined by the type of action being performed by the RCST. Each burst type is detailed in the following sections.

2.2.2.1. Traffic Bursts

Traffic bursts are used to carry useful data from the RCST to the gateway. EN 301 790 details traffic burst formatting, channel coding, and guard interval offsets. EN 301 790 defines two types of return traffic bursts, ATM cells and MPEG2-TS.

The SBS shall primarily support Internet Protocol (IP) over Multi-Protocol Encapsulation (MPE) over MPEG-2 Transport Streams (TS) protocol encapsulation on the forward link as described in Figure I - 5. Section packing shall be supported to reduce overhead for certain MPEG2-TS frame size.

The SBS shall support IP over ATM cell encapsulation (Figure I - 6) or IP over MPEG2-TS (Figure I - 7) for the Return Link TRF bursts. The RCST shall be able to support both ATM and MPEG encapsulation in order to be able to communicate with any hub operating in either ATM or MPEG mode on the return link.

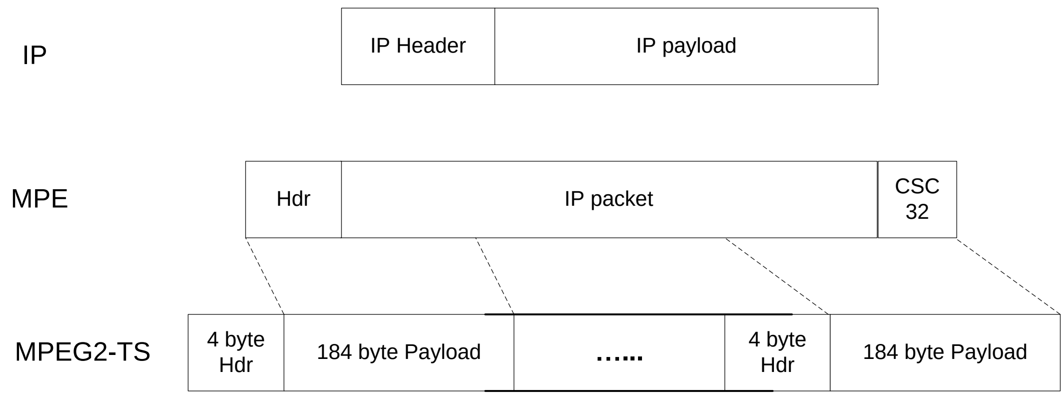

Figure I - 5 IP over MPE over MPEG-2-TS encapsulation on the Forward Link

The payload of an ATM traffic burst is composed of Natm concatenated ATM cells (Figure I - 6), each of length 53 bytes, plus an optional Np,atm byte prefix. ATM cells follow the structure of an ATM cell but do not necessarily support ATM classes of service.

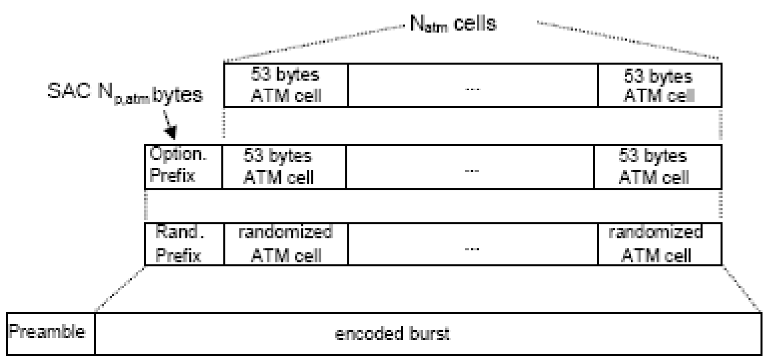

Figure I - 6 IP over ATM on the Return Link

Each traffic burst contains N MPEG concatenated 188-byte MPEG-2 TS packets (Figure I - 7) that are randomized and coded according to Section 2.2.4. The RCST indicates its ability to support MPEG traffic bursts to the NCC during logon.

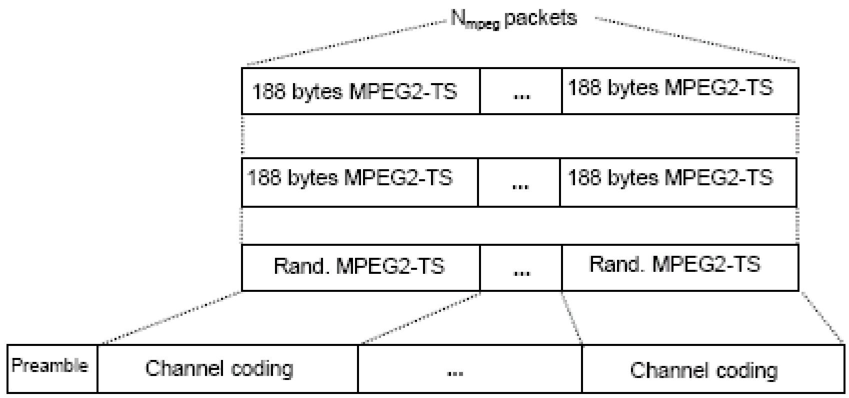

Figure I - 7 MPEG2-TS burst on the Return Link

2.2.2.2. Acquisition Bursts

Acquisition bursts (Figure I - 8) can be used by the RCST during network logon to achieve synchronization.

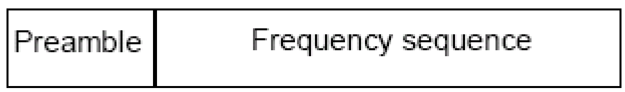

Figure I - 8 Composition of an ACQ burst

2.2.2.3. Synchronization Bursts

Synchronization bursts (Figure I - 9) are periodically used by the RCST to maintain synchronization with the network. These bursts may also contain a Satellite Access Control (SAC) field to pass MAC traffic, such as resource requests, to the gateway.

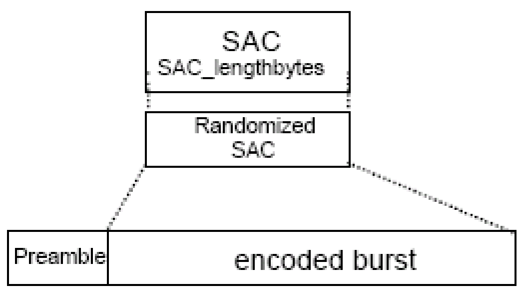

Figure I - 9 Composition of a SYNC burst

2.2.2.4. Common Signaling Channel Bursts

Common Signaling Channel bursts (Figure I - 10) are used by the RCST only during logon to identify itself to the NCC. The CSC burst contains RCST identification information as well as a description of capabilities supported.

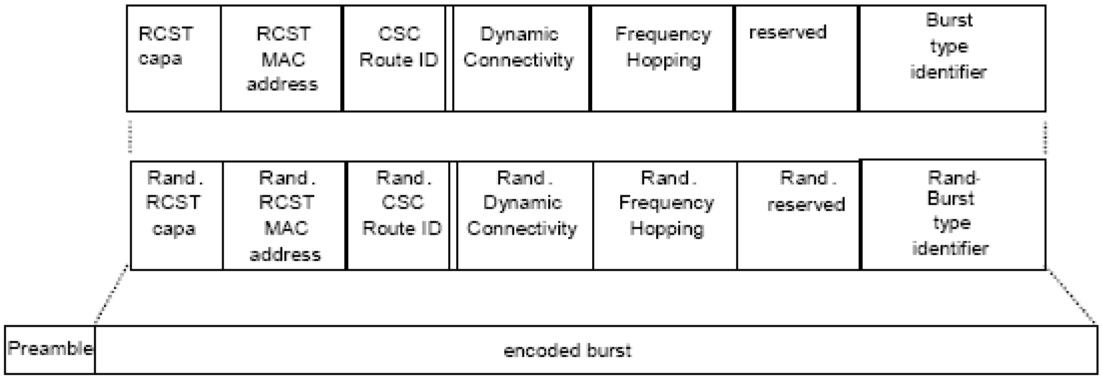

Figure I - 10 Composition of a CSC burst

2.2.3. Energy Dispersal

In order to comply with ITU Radio Regulations and ensure adequate binary transitions, all bursts shall be randomized according to EN 301 790 section 6.3.

2.2.4. Channel Coding

Channel coding will be applied to traffic and MAC bursts to provide a measure of error protection. Two types of coding will be used in the SBS system: Cyclical Redundancy Checking (CRC) and Turbo coding.

2.2.4.1. Cyclical Redundancy Checking

A CRC-16 code will be applied to all CSC and SYNC bursts utilizing the contention-based mini-slot access method. The NCC shall indicate to RCSTs through the Timeslot Composition Table (TCT) that CRC is to be applied to those bursts. CRC-16 shall be calculated on the randomized burst bit stream prior to any other coding as specified in EN 301 790 section 6.4.1

2.2.4.2. Turbo Coding

Turbo coding, as defined in EN 301 790 clause 6.4.4, shall be applied to all TRF, SYNC, and CSC bursts. Parameters of the turbo code, such as ordering and puncturing, shall be signaled to the terminal through the Timeslot Composition Table (TCT).

2.2.5. Modulation

Bursts shall be modulated using Quadrature Phase Shift Keying (QPSK).

2.2.5.1. QPSK Modulation

EN 301 790 section 6.5 describes the process for bit-mapping into the QPSK constellation. The process is summarized below.

The system shall employ a conventional Gray-coded QPSK modulation with absolute mapping (no differential encoding) and baseband shaping as defined in the Modulation and Baseband Shaping sections of Annex H. A configurable preamble shall be provided to the RCST through the TCT. Following preamble insertion, outputs C1 and C2 of the encoder shall sent to the QPSK bit mapper to be mapped to the I and Q channels, respectively, as shown in Figure I - 11. Figure I - 12 depicts bit mapping of the channels to the QPSK constellation.

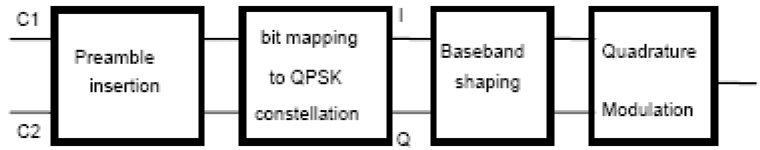

Figure I - 11 Processing after the encoder

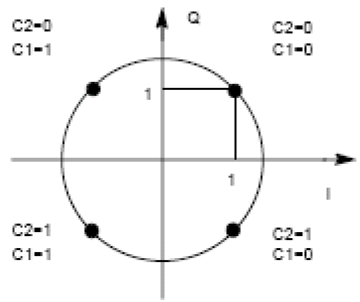

Figure I - 12 QPSK Constellation

2.3. Medium Access Control Messages

Section 6.6 of EN 301 790 details the methods of delivering MAC messages using Satellite Access Control (SAC) fields or Data Unit Labeling Method (DULM). The SBS system shall support these methods as applied to SYNC and MPEG-2 TRF bursts, briefly described below.

2.3.1. SAC Fields

MAC messages may be passed via SAC fields in SYNC and MPEG-2 TRF bursts. The SAC field may contain capacity requests or other M&C information from the RCST to the gateway. Table 8 in EN 301 790 details the syntax of the SAC field.

Configuration of the SAC field in SYNC bursts is determined by the parameters provided in the TCT. SAC fields contained in MPEG-2 TRF bursts are signaled directly by the RCST within the MPEG-2 TS packet adaptation field.

2.3.1.1. Mini-Slot Method

RCSTs are periodically assigned burst slots that are smaller than traffic time slots for use in maintaining synchronization with the network. As illustrated in Section 2.2.2.3, these bursts may contain a SAC field of length specified in the TCT. This SAC field may be used by the RCST to pass capacity requests or M&C traffic to the gateway.

Mini-slots may be contention-based and accessible by a group of RCSTs. In this event the SAC request is supported by the Group_ID and Logon_ID that uniquely identify a particular RCST on the network.

2.3.1.2. MPEG Adaptation Field Method

SAC messages may optionally, subject to gateway support, be carried in the adaptation field of the MPEG-2 TS header during TRF bursts. This method is not signaled by the NCC in the TCT, but is added by the RCST to TRF bursts as required. The format of a TS carrying a SAC field in this manner shall conform to ISO/IEC 13818-1 clause 2.4.3. This requires that the adaptation_field_control parameter be set to “10” when a SAC field is sent with no accompanying data payload, or "11" when the SAC is followed by a payload section. The syntax of the SAC adaptation field is described in EN 301 790 Table 12.

2.3.2. Data Unit Labeling Method

The RCST may send capacity requests and other M&C messages in the payload of TRF bursts using a dedicated M&C virtual channel designated by the NCC. This channel shall be signaled by the NCC to the RCST during logon and shall be assigned a special PID to distinguish it from the virtual channel for return data. The RCST may use this channel to pass MAC traffic using a time slot already assigned through the Terminal Burst Time Plan (TBTP), or the NCC may request status information and allocate time slots to an RCST.

2.3.3. ACM Fields

SBS systems that employ DVB-S2 on the forward link with the ACM option shall support the ACM sub-field as described in EN 301 790 clause 6.6.1.1. These sub-fields allow the RCST to pass link quality and requested modulation and coding information back to the gateway.

2.4. (NU) RETURN LINK RESOURCE SEGMENTATION

The aggregate capacity of the all satellite return link resources shall be divided among sets of RCSTs into superframes, each delineating a portion of time and frequency of the return link. A Superframe ID shall be assigned to blocks of consecutive superframes that are continuous in time, as illustrated in Figure I - 13, to identify the return link resources accessed by a given set of RCSTs.

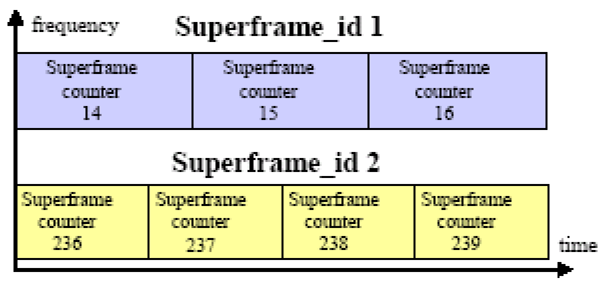

Figure I - 13 Example of superframes on separate set of frequencies

Superframes shall be segmented into frames and timeslots, where a superframe is composed of frames, and frames are composed of timeslots. The segmentation of resources among superframes, frames, and timeslots, and the specific parameters of each are left to the discretion of the system integrator, subject to the limitations specified in EN 301 790.

In a frame, timeslots shall be numbered from lowest frequency, first in time, to highest frequency, last in time. Each timeslot shall be uniquely identified by its superframe ID, superframe counter, frame number, and timeslot number

2.4.1. Multiple Frequency Time Division Multiple Access

Multiple-Frequency Time Division Multiple Access (MF-TDMA) allows a group of RCSTs to communicate with the hub using multiple carrier frequencies, each of which is divided into time slots. The NCC at the hub shall allocate to each active RCST a series of bursts, each defined by a frequency, a bandwidth, a start time and duration. Either Fixed-Slot MF-TDMA or Dynamic-Slot MF-TDMA can be employed.

In Fixed-Slot MF-TDMA the bandwidth and duration of successive traffic slots used by an RCST is fixed, as illustrated in Figure I - 14 where the arrow indicates a typical sequence of slots assigned by the NCC to one RCST. In this case, the TCT parameters of a superframe are fixed. A fixed MF-TDMA RCST can send a mix of SYNC and single size TRF bursts provided that the burst parameters fulfill the previous requirement. If the NCC requests a change in these parameters, then they will apply to a new superframe with a delay.

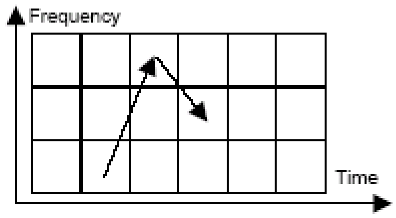

Figure I - 14 Fixed slot MF-TDMA

Dynamic-Slot MF-TDMA uses additional RCST flexibility to vary the bandwidth and duration of successive slots allocated to an RCST. In addition to changing carrier frequency and burst duration, the RCST may also change transmission rate and coding rate between successive bursts. The advantage of the more flexible RCST is more efficient adaptation to the widely varying transmission requirements typical of multimedia. The basic principle of the flexible RCST is illustrated in Figure I - 15, where the arrows show an RCST using successive slots with different bandwidths and durations.

The choice of Fixed-Slot or Dynamic-Slot MF TDMA is left to the discretion of the system integrator. Dynamic-Slot MF-TDMA is recommended in SBS systems because of its flexibility, efficiency, and security advantages when combined with link-layer encryption.

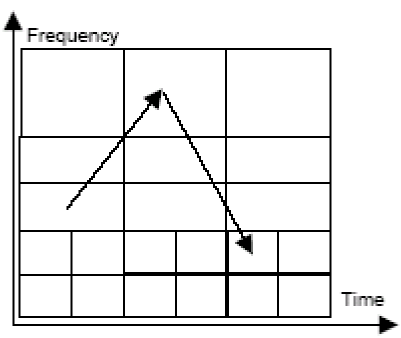

Figure I - 15 Dynamic slot MF-TDMA

2.4.2. (NU) Capacity Request Categories

The following capacity request categories shall be used, as defined in EN 301 790:

Continuous Rate Assignment (CRA) – CRA capacity shall be continuously provided to the RCST by the NCC through the RCST's active session. CRA should be used for traffic that requires a fixed guaranteed bit rate such as application data that has low latency and jitter tolerance.

Rate Based Dynamic Capacity (RBDC) – The RCST shall request RBDC to satisfy time or delay-sensitive bandwidth requirements as needed above allocated CRA. The NCC shall automatically set RBDC allocations to zero after a time specified in the NCC configuration to prevent problems associated with a hanging request. RBDC capacity shall not be provided to an RCST that has reached its maximum allowed capacity assignment for the superframe.

Volume Based Dynamic Capacity (VBDC) – The RCST shall request VBDC to satisfy routine bandwidth requirements as needed above allocated CRA. The NCC shall sum the total of all VBDC requests issued by a particular RCST, subtracting any VBDC capacity that is allocated to the same. VBDC shall be allocated only if all CRA and RBDC capacity has been assigned to the RCST population, unused capacity is available, and the RCST maximum allocation has not reached during the superframe period.

Absolute Volume Based Dynamic Capacity (AVBDC) – The RCST may issue an AVBDC request to explicitly request the total VBDC it requires in the event the integrity of aggregate VBDC request information at the NCC is in question. Upon receiving an AVBDC request from an RCST, the NCC shall discard the currently held VBDC sum for the RCST and use the information in the AVBDC request..

Free Capacity Assignment (FCA) – After assigning CRA, RBDC, and VBDC capacity, the system may assign any unused capacity to RCSTs that have not met their maximum capacity allotment in a method determined by the system integrator. RCSTs may not request FCA from the NCC, nor shall a FCA be guaranteed.

2.5. CONTROL AND MANAGEMENT

This section describes the basic control and management functions of the DVB-RCS network necessary for the RCST to determine system parameters, relay its capabilities to the NCC, and pass data using common protocols. Clause 8 of EN 301 790 details the control and management functions of the DVB-RCS system. The sections below briefly describe these functions and provide clarification on the functions that shall be implemented in the SBS system.

2.5.1. Protocol Stack Model

On the return link the protocol stack is based on ATM cells or optional MPEG2-TS packets mapped onto TDMA bursts. For transmission of IP datagrams, the protocol stacks used on the return link are as follows:

- ATM based return link: IP/AAL5/ATM
- Optional MPEG return link: multiprotocol over MPEG2 Transport Streams encapsulation

In the forward link the protocol stack is based on the DVB/MPEG2-TS standard (TR 101 154). For transmission of IP datagrams, the protocols stacks used on forward link are as follows:

- multiprotocol encapsulation over MPEG2 Transport Streams, as defined in [2] and [5]

The return link of the SBS DVB-RCS system shall be based on ATM cells (standard) or MPEG2-TS Optional. IP datagrams on the return link shall be encapsulated via MPE before being mapped into MF-TDMA traffic bursts as defined in ETSI TR 101 202 and EN 301 192

For ATM cells, the 53 bytes of the ATM cell are made of 5 bytes of header and 48 bytes of payload. ATM cells are used either for user traffic or for control and/or management traffic (handled by upper layers). The 5-byte header of an ATM cell shall follow the ATM UNI cell format

For the optional MPEG2-TS the IP datagrams on the return link shall be encapsulated via MPE before being mapped to MPEG-2 TS packets as defined in ETSI TR 101 202 and EN 301 192

An example protocol stack for user traffic is illustrated in Figure I - 16.

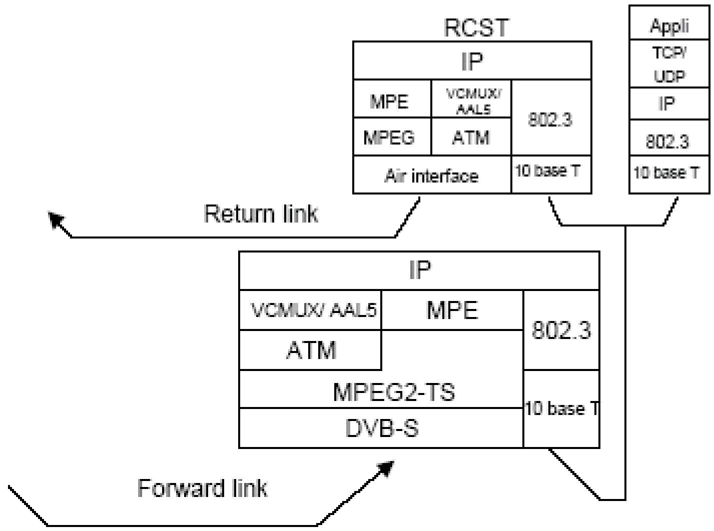

Figure I - 16 Protocol Stack for user traffic with RCST (Forward and Return Link)

2.5.2. Forward Link Signaling

The SBS system shall support the set of Service Information (SI) tables that provide information regarding the broadcast network as defined in EN 300 468 and clause 8.3 of EN 301 790. These tables carry forward link signaling that describes the overall forward and return link network as well as event data such as superframe construction and burst plans.

2.5.2.1. General SI Tables

General SI data that describes the network organization and structure are contained in six types of table. These tables are broadcast over the forward link in the form of private data structured according to IOS/IEC 13818-1. The tables include the following:

- **Superframe Composition Table (SCT) –** Describes the subdivision of the satellite interactive network into superframes and frames.
- **Frame Composition Table (FCT) –** Describes partitioning of frames into time slots.
- **Timeslot Composition Table (TCT) –** Describes transmission parameters for each timeslot type identifier.
- **Satellite Position Table (SPT) –** Contains satellite ephemeris data used to periodically update terminal burst position.
- **Correction Message Table (CMT) –** Data sent from NCC to RCSTs for the purpose of burst timing, frequency, and power adjustments.
- **Terminal Burst Time Plan (TBTP) –** Contains time slot assignments for each RCST.

2.5.2.2. Multicast Mapping Table

The Multicast Mapping Table (MMT) provides associations between multicast IP addresses and PIDs used to carry multicast data. Using the MMT, IP encapsulators can dynamically update multicast-to-PID map data and provide the associations to all RCSTs through forward link signaling. SBS systems shall support the MMT as specified in the SatLabs System Recommendations appendix and described below.

The most recently transmitted version of the multicast_to_PID_map section with the current_next_indicator set to a value of '1' shall always apply to the current data within the TS. Any changes to the multicast sessions carried within the TS shall be described in an updated version of the MPEG MMT carried in Transport Stream. A new version of a multicast_to_PID_map becomes valid when the last byte of that section with a new version_number and with the current_next_indicator set to ‘1’ is processed. Table I - 1 contains the structure of the MMT fields.

Table I - 1 Multicat Mapping Table Structure

Syntax	No. of bits	Notes
Multicast_to_PID_map {		
table_id	8	'0xC0 '
Section_syntax_indicator	1	'1'
Private_indicator	1	
reserved	2	'11'
Private_section_length	12	
table_id_extension	16	
reserved	2	
Version_number	5	
Current_next_indicator	1	
Section_number	8	
last_section_number	8	
for (i=0; I<N; i++) {		
IPv6_flag	1	
reserved	2	
elementary_PID	13	
for (I=0; i<4; i++) {		
IPv4_address	8	
}		
If (IPv6_flag == '1') {		
for (i=0; i<6; i++) {		
IPv6_address	16	
}		
}		
}		
CRC_32	32	
}		

The fields of the MMT are defined as follows:

- **table_id:** an 8-bit field which identifies the MPEG Private Table this section belongs to. Here, the value is 0xC0.
- **section_syntax_indicator:** a 1-bit indicator. When set to ‘1’, it indicates that the private section follows the generic section syntax beyond the private_section_length field. When set to ‘0’, it indicates that the private_data_bytes immediately follow the private_section_length field.
- **private_indicator:** a 1-bit user definable flag; it is not used
- **reserved:** these bits may be used in the future; all reserved bits shall be set to '1'
- **private_section_length:** a 12-bit field specifying the number of remaining bytes in the section immediately following the private_section_length field up to the end of the private_section. The value in this field shall not exceed 4093 (0xFFD).
- **table_id_extension:** a 16-bit field; it is not used.

- **version_number:** a 5-bit field that defines the version number of the private_section. The version_number shall be incremented by 1 modulo 32 when a change in the information carried within the private_section occurs. When the current_next_indicator is set to '0', then the version_number shall be that of the next applicable private_section with the same table_id and section_number.
- **current_next_indicator:** a 1-bit field, which when set to '1' indicates that the private_section sent is currently applicable. When the current_next_indicator is set to '1', then the version_number shall be that of the currently applicable private_section. When the bit is set to '0', it indicates that the private_section sent is not yet applicable and shall be the next private_section with the same section_number and table_id to become valid.
- **section_number:** an 8-bit field that gives the number of the private_section. The section_number of the first section in a private table shall be 0x00. The section_number shall be incremented by 1 with each additional section in this private table.
- **last_section_number:** an 8-bit field that specifies the number of the last section (that is, the section with the highest section_number) of the private table of which this section is a part.
- IPv6_flag: a one-bit field that specifies whether IPv6 addressing is used. When set to '0', it indicates that IPv4 (32-bit address space) addressing is used.
- elementary_PID: 13-bit field specifying the PID of the Transport Stream packets which carry the associated multicast data
- IPv4_address: a 32-bit field that specifies the multicast IP address for data carried in the elementary_PID. The IP address is fragmented into four fields of 8 bits, where IPv4_address[0] containing the most significant byte of the IP address, while IPv4_address[3] contains the least significant byte of the IP address.
- IPv6_address: a 96-bit field that specifies the multicast IP address for data carried in the elementary_PID. The IP address is fragmented into six fields of 16 bits, where IPv6_address[0] containing the most significant 16 bit field of the IP address, while IPv6_address[5] contains the least significant 16 bit field of the IP address.
- Together with IPv4_address, IPv6_address will yield the following IP address format: XX:XX:XX:XX:XX:XX:d.d.d.d, where "XX" is the hexadecimal value of the high-order 16-bit pieces of the address and "d" is the decimal value of the low-order bit pieces of the address. [RFC2373].
- CRC_32: a 32-bit field that contains the CRC value that gives a zero output of the registers in the decoder defined in Annex B of EN 300 468 after processing the entire private section.

2.5.2.3. RCS Map Table

SBS systems shall support the RCS Map Table (RMT) as described in clause 8.5.5.12 of EN 301 790. The RMT links different RCST populations to a specific Forward Link Service (FLS). RCSTs select the appropriate FLS by searching the RMT for a matching Population ID in the linkage descriptors. Each corresponding FLS should carry the set of signaling tables as described in Section 2.8.2.

2.5.2.4. Transmission Mode Support Table

Systems that employ DVB-S2 with ACM on the forward link, as described in Annex J, shall support the Transmission Mode Support Table (TMST) defined in clause 8.5.5.13 of EN 301 790. The TMST defines the DVB-S2 transmission modes (modulation and coding rate) that are supported by the network for forward link transmissions. This table will enable RCSTs to select supported modes of operation when composing ACM MODCOD_RQ messages.

2.6. (NU) SYNCHRONIZATION PROCEDURES

The RCST shall conform to the synchronization procedures detailed in EN 301 790. These procedures define required and optional steps used by the RCST to logon to the network, maintain synchronization with the network during a session, and logoff of the network. The general procedures are described below.

2.6.1. RCST States and State Transition

The RCST shall support the following states as described in EN 301 790 clause 7.1:

- **Hold** – The RCST is in hold mode
- **Inactive Off/Stand-by** – The RCST is powered off or has lost synchronization
- **Receive sync** – The RCST has acquired the forward link
- **Ready for coarse sync** – The RCST has been detected by the NCC and may initiate a coarse synchronization procedure
- **Ready for fine sync** – The RCST has been detected by the NCC and may initiate a fine synchronization procedure
- **Fine sync** – The RCST is synchronized and can send traffic

The RCST shall transition between states according to the method illustrated in Figure I - 17.

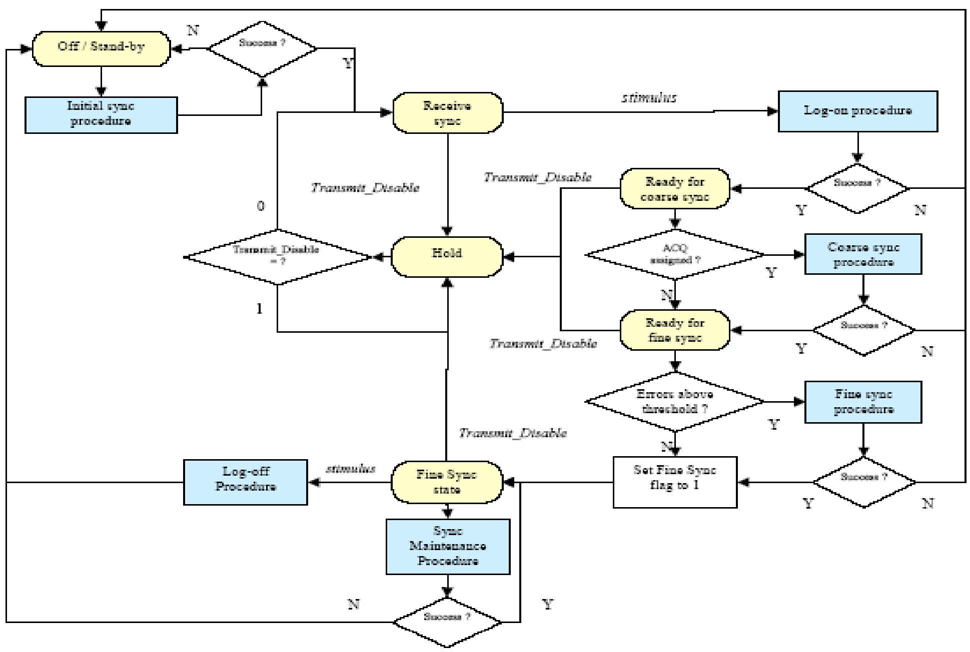

Figure I - 17 RCST Synchronization state diagram

2.6.2. Session Synchronization

Terminals shall initialize, maintain, and close sessions with the gateway according to the procedures defined in EN 301 790 clause 7. The process is illustrated in Figure I - 18 and described in the following sections.

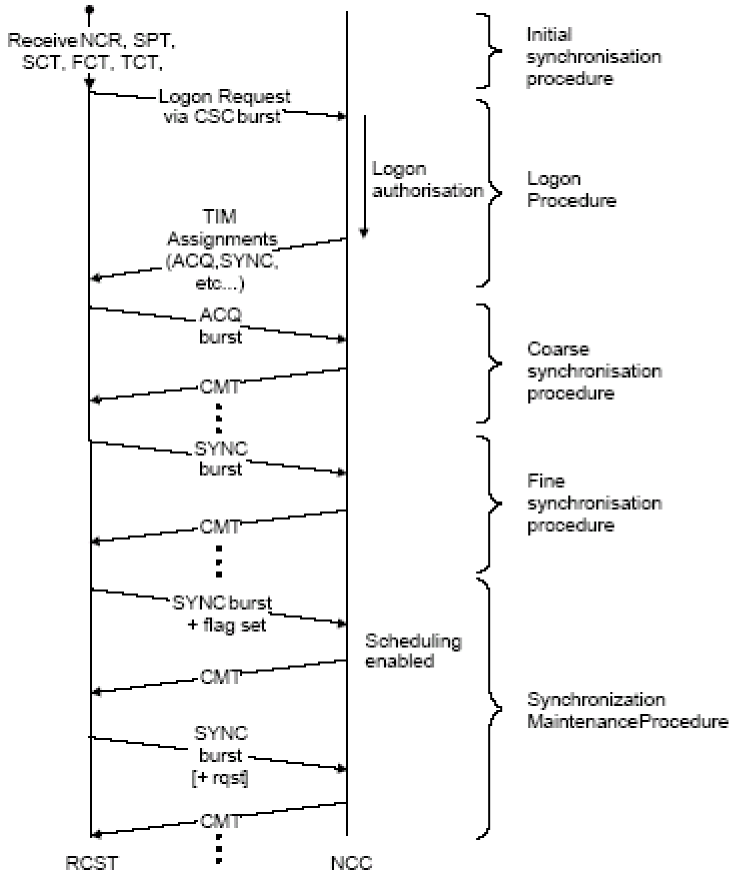

Figure I - 18 Example of RCST Signaling During Logon, Maintenance

2.6.2.1. Initial Synchronization

The RCST shall initialize its internal clock by tracking the NCR broadcast by the NCC over the broadcast channel. This information shall be used to calculate the range to the satellite and propagation delay for both the forward and return links.

2.6.2.2. Logon

The RCST shall request access to the network. The NCC shall accept or reject the request and follow with a TIM containing synchronization ACQ and SYNC timeslot assignments and other network information as described in Section 2.5.2.

2.6.2.3. Coarse Synchronization

The NCC may require the RCST to undergo coarse synchronization using ACQ bursts to improve its physical synchronization. In a network where the NCC can correct all frequency and timing errors other than differential Doppler between the RCST and the NCC, the RCST does not need to perform the coarse synchronization procedure. If RCST does not receive a TIM from the NCC containing an ACQ assign descriptor, it may enter the Fine Synchronization Procedure.

2.6.2.4. Fine Synchronization

The NCC may require the RCST to undergo fine synchronization using SYNC bursts to complete its physical synchronization.

2.6.2.5. Synchronization Maintenance

The RCST shall maintain physical synchronization during the entire session using SYNC bursts as assigned by the NCC. The NCC shall issue SYNC bursts as necessary to maintain network synchronization according to parameters established by the system integrator.

2.6.2.6. Logoff

When a logoff instruction is initiated as a result of session termination (normal) or failure (abnormal), the RCST shall cease all active transmissions and transition to the Stand by state. The NCC shall remove logical address and SYNC timeslot references for the terminal and make them available for other terminals.

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ANNEX J: DVB-S2 (DVB-SATELLITE VERSION 2)

1. (NU) INTRODUCTION

This annex describes the process of second generation DVB modulation and channel coding (DVB-S2), as defined in ETSI EN 302 307. The SBS may use this modulation and coding scheme for data broadcast channels in lieu of or in addition to the DVB-S scheme described in Appendix H.

1.1. (NU) AIM

This annex summarizes the details of the DVB-S2 Modulation process as described in EN 302 307.

1.2. (NU) GENERAL

Figure J-1 highlights the portion of the SBS that is described in this annex.

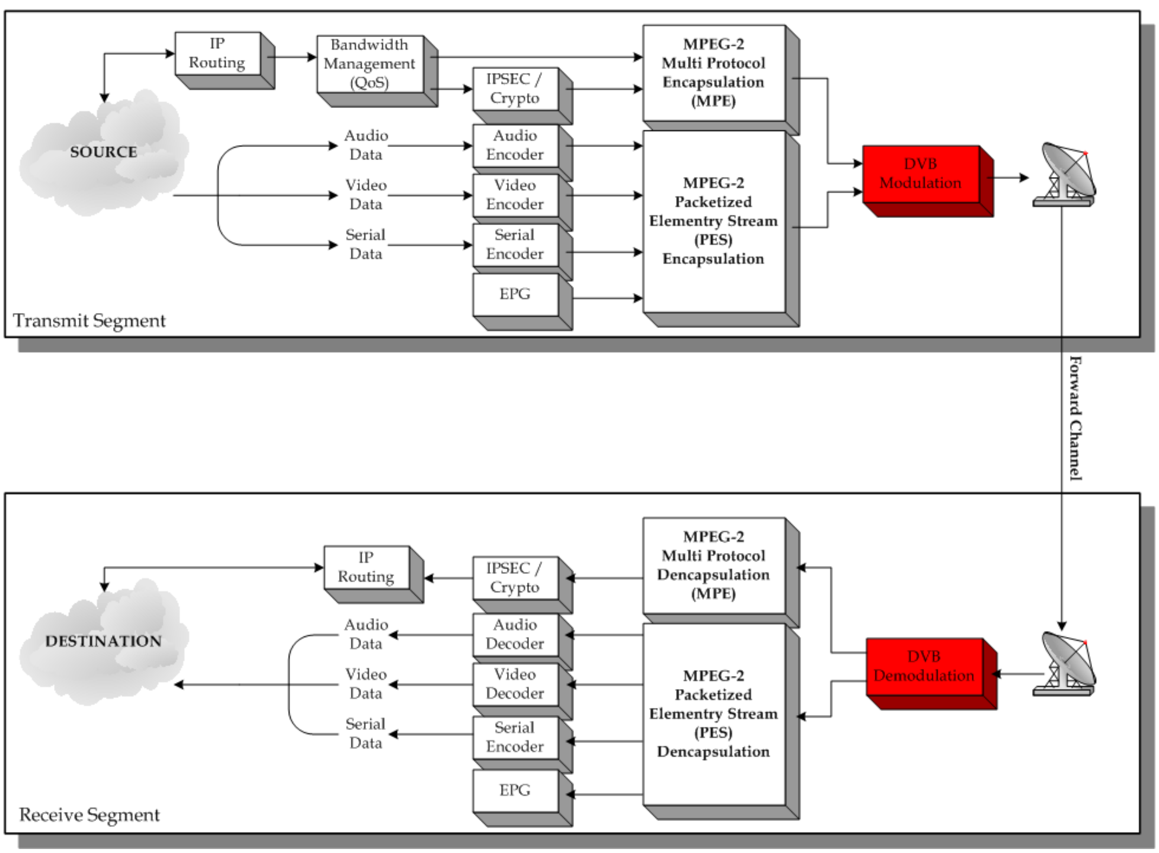

Figure J - 1 DVB Modulation Conceptual Overview

2. (NU) SECOND GENERATION DVB MODULATION

The DVB-S2 coding and modulation standard (EN 302 307) was formally ratified by ETSI in 2005. Drawing from the success of earlier schemes such as DVB-S and DVB-Digital Satellite News Gathering (DSNG), DVB-S2 specifies a rich set of features that are suitable for broadcast, professional, and interactive services. The standard includes high-order modulations, an efficient coding scheme, and advanced differentiated services that will benefit systems through bandwidth savings, reduced terminal size, and improved reliability.

Forward channel data sent over the SBS utilizing the waveform described in this annex will be transmitted according to the DVB-S2 standard. Figure J-2 illustrates a functional block diagram of the DVB-S2 system. The system accepts data from a single or multiple input streams which is sliced into data fields for processing. The fields are scrambled, Forward Error Correction (FEC) encoded, bit-mapped, encapsulated into physical layer frames, and finally modulated for transmission.

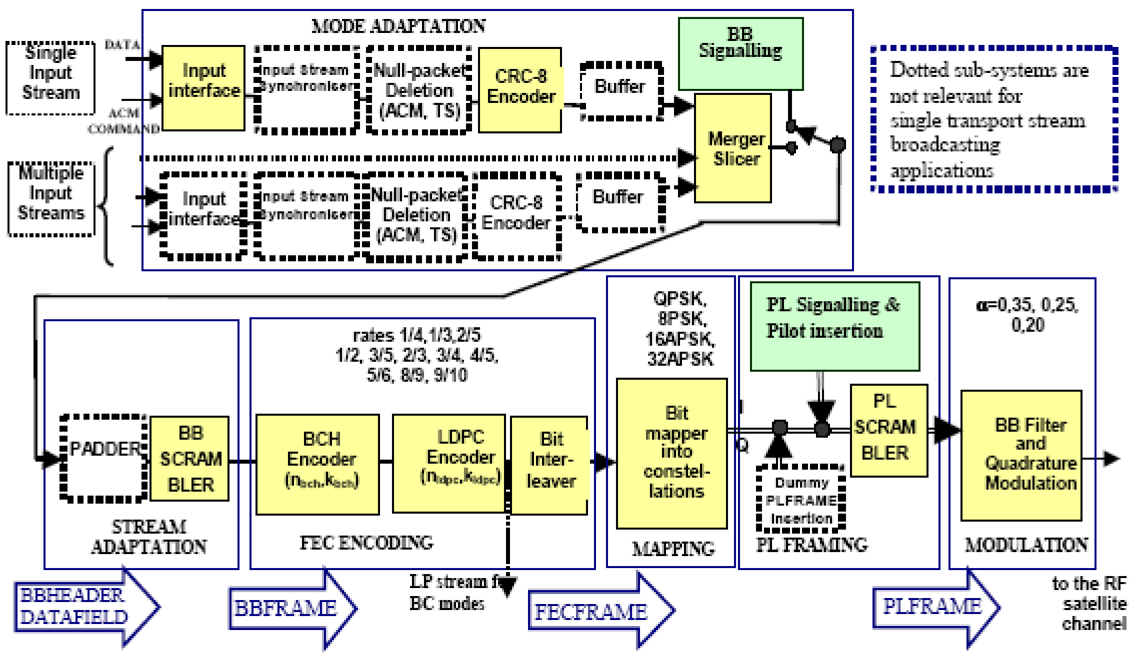

Figure J - 2 DVB-S2 System Block Diagram

2.1. (NU) MODE ADAPTATION

Mode adaptation of input data shall be performed in accordance with Section 5.1 of EN 302 307. This process is described below.

The mode adaptation subsystem performs the functions of input interfacing, input stream synchronization, null packet deletion, CRC encoding, merging and slicing of input streams, and baseband header insertion. Stream synchronization, null packet deletion, CRC encoding, and merging functions are dependant upon the type of data provided (generic stream or MPEG-2 TS) and number of input streams, and may not be performed in all cases. Input to the mode adaptation subsystem can be in the form of a single or multiple streams, with acceptable stream types including MPEG-2 TS, generic continuous bit stream, or generic packetized stream. Output of the subsystem is a DATA FIELD prepended by a baseband signaling header.

Single MPEG-2 TS systems use for output of a CCM broadcast signal will CRC-8 encode and map TS packets to FEC frames for broadcast. The data will be uniformly encoded, modulated, and broadcast. For systems employing the ACM scheme, the mode adaptation field will add modulation and encoding information to the baseband header prepended to the DATA FIELD. This information will notify the receiver of mode adaptation settings and input stream format for proper demodulation and decoding.

SBS systems shall be mainly concerned with input streams of the MPEG-2 TS type, with IP data encapsulated using the MPE process in Annex D. The MPEG-2 TS shall be formatted according to ISO/IEC 13818-1.

2.1.1. Input Stream Synchronizer

The optional Input Stream Synchronizer (ISS) subsystem shall provide a suitable means to guarantee Constant Bit Rate (CBR) and constant end-to-end transmission delay for packetized input streams. The ISS mitigates potential variable transmission delay on user information caused by data processing in the DVB-S2 modulator. If implemented, the ISS shall follow the specification according to EN 302 307 Annex D.

2.1.2. Null-Packet Deletion

Null-packet deletion is relevant only for ACM modes in systems using the MPEG-2 TS input format. In ACM modes, MPEG-2 TS null packets (PID=8191_D) shall be identified and removed. A counting mechanism shall be used to allow the receiving demodulator to replace the removed packets and accurately reproduce the MPEG-2 TS. This function shall follow the process specified in EN 302 307 Annex D.

2.1.3. CRC-8 Encoding

The DVB-S2 system shall process packetized input streams through a systematic 8-bit CRC encoder in accordance with EN 302 307 clause 5.1.4. The result of the CRC-8 encoding process shall replace the packet sync byte so that the sync byte of the N+1th packet contains the CRC-8 result of the Nth packet. The sync byte shall be stored in the baseband header according to Section 2.2. Continuous streams shall pass through the CRC-8 subsystem without modification.

2.1.4. Merger/Slicer

The merger/slicer subsystem shall operate according to EN 302 307 clause 5.1.5. This subsystem buffers input streams, slices individual streams into appropriately-sized DATA FIELDs, and performs stream multiplexing by merging the slices onto a single output. The merging function shall not apply for systems containing a single input stream.

The slicer shall read from one of its input streams and form a DATA FIELD with a length dependant on the system application and FEC mode. The system may read a fixed number of bits or, in the case of packetized streams, an integer number of User Packets (UP). Prioritization policies are application dependant and shall follow strategies described in EN 302 307 Table 4 and Table D.2.

If a DATA FIELD is not available on request of the subsystem on any data port, the Physical Layer Framing subsystem shall generate and send a dummy Physical Layer Frame (PLFRAME).

Systems using packetized input streams shall record the number of bits from the beginning of the DATA FIELD to the beginning of the first UP in the DATA FIELD in a Sync Distance (SYNCD) field. This value will enable the receiver to recover UP synchronization upon demodulation and decoding of the stream.

2.1.5. Baseband Header Insertion

The system shall prepend a 10 byte baseband header (BBHEADER) to the DATA FIELD in accordance with EN 302 307 clause 5.1.6. The BBHEADER contains signaling information regarding the input stream type and functions in use. The sync byte for packetized streams, that was displaced by the CRC-8 value from Section 2.1.3, shall be stored in the BBHEADER for restoral at the receiver.

2.2. (NU) STREAM ADAPTATION

Stream adaptation includes padding of the DATA FIELD to achieve a constant length baseband frame (BBFRAME). The BBFRAME consists of the baseband header (BBHEADER), DATA FIELD, and any padding if required. Randomization of the field is accomplished using a Pseudo Random Binary Sequence (PRBS).

2.2.1. Padding

Padding shall be applied, if required, to the BBFRAME according to EN 302 307 clause 5.2.1. The padding shall be appended to the combined BBHEADER and DATA FIELD so that the total length including padding fills a Bose-Chaudhuri-Hocquenghem (BCH) encoded block for the mode in use. Section 2.3 describes the FEC process.

2.2.2. Baseband Scrambling

The completed BBFRAME shall be randomized according to EN 302 307 clause 5.2.2. Randomization shall be accomplished using a pseudo-random binary sequence and shall be synchronous with the BBFRAME.

2.3. FEC ENCODING

The DVB-S2 system uses a concatenation of BCH block outer coding with Low Density Parity Check (LDPC) inner coding. FEC encoding is applied to an input stream of BBFRAMEs resulting in an output stream of FEC-encoded frames (FECFRAMES).

Depending on the application, DATA FIELDs are sliced so that the final LDPC coded block is either 16200 bits or 64800 bits. BCH coding is applied to the BBFRAME, and the resulting BCH-encoded block undergoes LDPC encoding at a specified rate. The LDPC code rate may be statically applied (CCM, VCM), or dynamically switched on a frame-by-frame basis (ACM). Available LDPC code rates range from 1/4 to 9/10. Table J-1 and Table J - 2 list the coding parameters for normal and short FECFRAMEs, respectively.

Table J - 1 Normal FECFRAME Coding parameters

LDPC code	BCH Uncoded Block K_{bch}	BCH coded block N_{bch} LDPC Uncoded Block k_{ldpc}	BCH t-error correction	LDPC Coded Block n_{ldpc}
1/4	16 008	16 200	12	64 800
1/3	21 408	21 600	12	64 800
2/5	25 728	25 920	12	64 800
1/2	32 208	32 400	12	64 800
3/5	38 688	38 880	12	64 800
2/3	43 040	43 200	10	64 800
3/4	48 408	48 600	12	64 800
4/5	51 648	51 840	12	64 800
5/6	53 840	54 000	10	64 800
8/9	57 472	57 600	8	64 800
9/10	58 192	58 320	8	64 800

Table J - 2 Short FECFRAME Coding Parameters

LDPC Code identifier	BCH Uncoded Block K_{bch}	BCH coded block N_{bch} LDPC Uncoded Block k_{ldpc}	BCH t-error correction	Effective LDPC Rate $k_{ldpc}/16\ 200$	LDPC Coded Block n_{ldpc}
1/4	3 072	3 240	12	1/5	16 200
1/3	5 232	5 400	12	1/3	16 200
2/5	6 312	6 480	12	2/5	16 200
1/2	7 032	7 200	12	4/9	16 200
3/5	9 552	9 720	12	3/5	16 200
2/3	10 632	10 800	12	2/3	16 200
3/4	11 712	11 880	12	11/15	16 200
4/5	12 432	12 600	12	7/9	16 200
5/6	13 152	13 320	12	37/45	16 200
8/9	14 232	14 400	12	8/9	16 200
9/10	NA	NA	NA	NA	NA

2.3.1. (NU) Interleaving

Interleaving shall be performed in accordance with Section 5.3.3 of EN 302 307. The interleaving process is described below.

Bit interleaving using a block interleaver shall be performed on the LDPC output for 8PSK, 16APSK, and 32APSK modes. Data is serially written into the interleaver column-wise, and serially read out row-wise. The block interleaver configuration for each modulation format is show in Table J - 3.

Table J - 3 Bit Interleaver Configuration

Modulation	Rows (for n_{ldpc} = 64 800)	Rows (for n_{ldpc} = 16 200)	Columns
8PSK	21 600	5 400	3
16APSK	16 200	4 050	4
32APSK	12 960	3 240	5

2.4. (NU) BIT MAPPING

Bit mapping shall be performed in accordance with Section 5.4 of EN 302 307. The bit mapping process is described below.

Each FECFRAME shall be serial-to-parallel converted and the resulting parallel sequence mapped into a constellation, generating a (I,Q) sequence of variable length depending on the modulation efficiency, η_{MOD}. From an input FECFRAME, a complex FECFRAME (XFECFRAME) shall be generated at the output composed of 64800/ η_{MOD} (normal FECFRAME) or 16200/ η_{MOD} (short FECFRAME) modulation symbols. Each modulation symbol shall be a complex vector in the format (I,Q) or equivalent format $\rho \exp(j\varphi)$.

2.4.1. QPSK Constellation

The system shall use a conventional Gray-coded QPSK modulation with absolute mapping (no differential encoding). Bit mapping to the QPSK modulation shall following Figure J-3. The normalized average energy per symbol shall be equal to ρ^2=1.

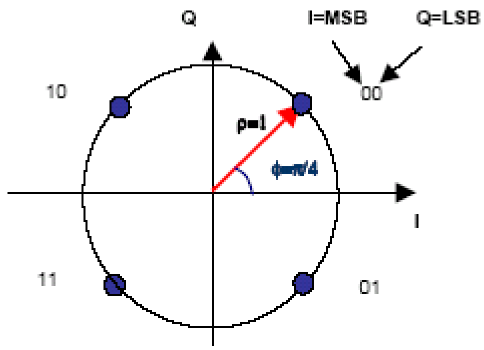

Figure J - 3 QPSK Constellation

2.4.2. 8PSK Constellation

The system shall use a conventional Gray-coded 8PSK modulation with absolute mapping (no differential encoding). Bit mapping to the 8PSK modulation shall following Figure J-4. The normalized average energy per symbol shall be equal to $\rho^2=1$.

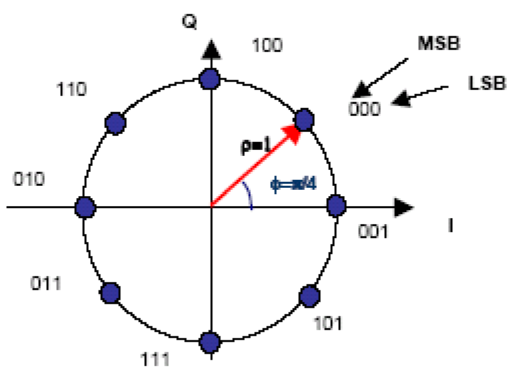

Figure J – 4 8PSK Constellation

2.4.3. 16APSK Constellation

The 16APSK constellation, shown in Figure J-5, shall be composed of two concentric rings of uniformly spaced 4 and 12 PSK points, respectively in the inner ring of radius R_1 and outer ring of radius R_2. The ratio of the outer ring radius to inner ring radius ($\gamma=R_2/R_1$) shall comply with EN 302 307. If $4[R_1]^2+12[R_2]^2=16$ then the average signal energy becomes 1.

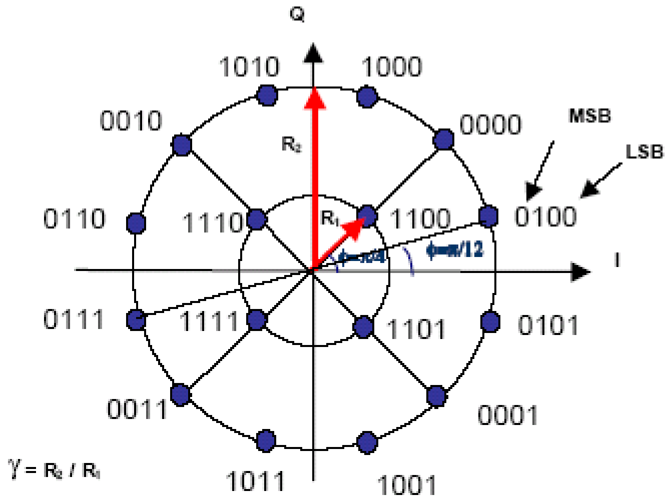

Figure J - 5 16APSK Constellation

2.4.4. 32APSK Constellation

The 32APSK constellation, shown in Figure J-6, shall be composed of three concentric rings of uniformly spaced 4, 12, and 16 PSK points, respectively in the inner ring of radius R_1, intermediate ring of radius R_2, and outer ring of radius R_3. The values $\gamma_1=R_2/R_1$ and $\gamma_2=R_3/R_1$ shall comply with EN 302 307. If $4[R_1]^2+12[R_2]^2+16[R_3]^2=32$ then the average signal energy becomes 1.

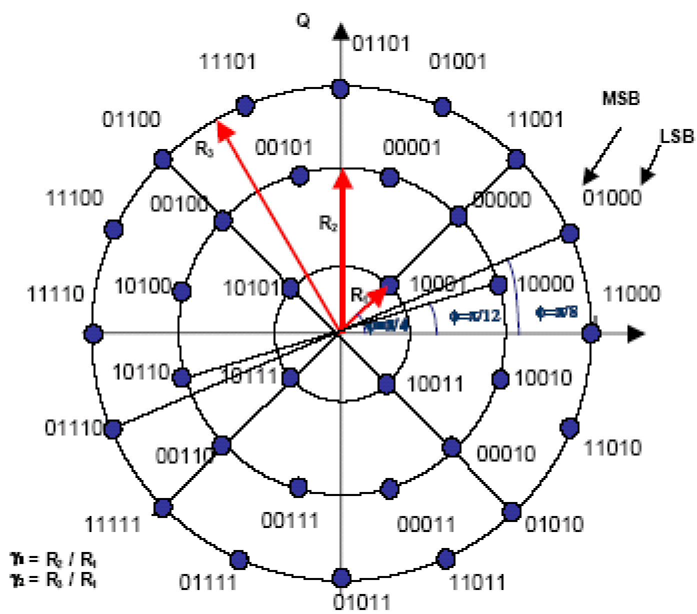

Figure J - 6 32APSK Constellation

2.5. (NU) PHYSICAL FRAMING

The physical framing subsystem synchronously maps XFECFRAMES to PLFRAMEs, prepending a physical layer header (PLHEADER) describing PLFRAME modulation and coding parameters to the receiver. Dummy PLFRAMEs are generated by the system when no XFECFRAME is ready for processing, and optional pilot symbols are added as required to assist the receiver with signal recovery. PLFRAMES are based on SLOTs of 90 modulation symbols with the number of SLOTs required for a PLFRAME dependant upon modulation efficiency, η_{MOD}, and use of pilot symbols as indicated in Table J - 4.

Table J - 4 PLFRAME SLOT Composition

	n_{ldpc} = 64 800 (normal frame)		n_{ldpc} = 16 200 (short frame)	
η_{MOD} (bit/s/Hz)	S	η % no-pilot	S	η % no-pilot
2	360	99,72	90	98,90
3	240	99,59	60	98,36
4	180	99,45	45	97,83
5	144	99,31	36	97,30

The PLHEADER is intended for receiver synchronization and physical layer signaling. It is Binary PSK (BPSK) modulated and consists of one SLOT (90 symbols) defining the following fields:

- **Start Of Frame (SOF)-** 26 symbols indicate the start of a PLFRAME
- **Physical Layer Signaling (PLS) code**- Non-systematic binary code of length 64 and dimension 7 with minimum distance d_{min}=32, equivalent to the first order Reed-Muller under permutation. It transmits 7 bits as follows: 1) A 5-bit MODCOD identifying the modulation and encoding, 2) A 2-bit TYPE field indicating the FECFRAME length and presence of optional pilot symbols.

Pilot symbols may be added to the PLFRAME to assist the receiver with synchronization and signal recovery. When enabled, a block of 36 pilot symbols shall be added to the PLFRAME every 16 SLOTs after the PLHEADER. Pilot symbols may be enabled on a frame-by-frame basis in VCM and ACM systems.

Prior to modulation the PLFRAME (excluding the PLHEADER) shall be scrambled for energy dispersal.

2.6. BASEBAND SHAPING AND QUADRATURE MODULATION

After randomization the signals shall be square root raised cosine filtered. Depending on the service requirements, the roll-off factor will be α=0.35, 0.25, or 0.20.

The baseband square root raised cosine filter shall have a theoretical function defined by the following expression:

$$H(f) = 1 \qquad |f| < f_N(1-\alpha)$$

$$H(f) = \left\{ \frac{1}{2} + \frac{1}{2}\sin\frac{\pi}{2 f_N}\left[\frac{f_N - |f|}{\alpha}\right]\right\}^{\frac{1}{2}} \qquad \text{for} \qquad f_N(1-\alpha)$$

$$H(f) = 0 \qquad |f| > f_N(1+\alpha)$$

Where: $f_N = \frac{1}{2T_S} = \frac{R_S}{2}$ is the Nyquist frequency and α is the roll-off factor.

3. ADDITIONAL OPERATING MODES

3.1. BACKWARDS COMPATIBILITY

DVB-S2 may be considered a successor to the current DVB-S standard (EN 300 421) and may be introduced for long-term migration. However, because of changes in modulation, coding, and physical layer framing structure, currently fielded receivers that conform to EN 300 421 will not be able to receive and decode DVB-S2 signals. SBS systems shall implement migration plans that allow interoperability with currently fielded equipment during the migration to the new standard. If implemented, DVB-S2 backwards compatibility modes shall be performed in accordance with Annex F of EN 302 307. The process is described below.

The DVB-S2 specification includes a backwards compatible mode that allows the system to transmit both a high-priority (HP) DVB-S stream (according to EN 300 421) and a low-priority (LP) DVB-S2 stream via a non-uniform 8PSK signal. The system, illustrated in Figure J-7, accepts a HP TS and performs encoding operations according to EN 300 421. In parallel, a second LP TS is accepted and follows the encoding and physical layer framing process according to EN 302 307. The resulting HP and LP streams are hierarchically mapped to a non-uniform 8PSK constellation, shaped, and modulated for transmission. At the end of the migration period the backwards compatibility mode may be disabled.

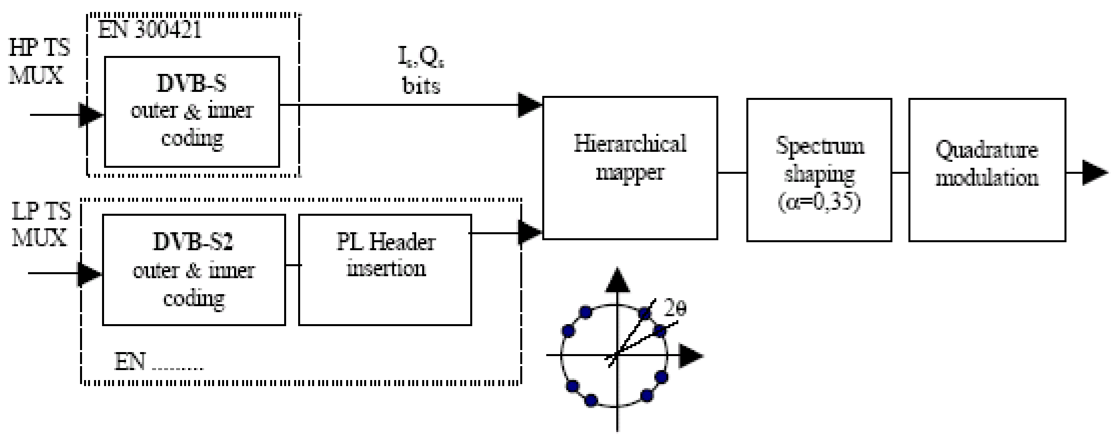

Figure J - 7 DVB-S2 Backwards-Compatibility Mode Signal Diagram

3.2. VARIABLE CODING AND MODULATION

The DVB-S2 may deliver differentially encoded and modulated broadcast services via multiple MPEG-2 TS. An example configuration is shown in Figure J-8. VCM could be used to provide differentially encoded services to a diverse set of terminal groups based on terminal group capabilities, terminal group link characteristics, or to provide robust protection to a critical data stream and less protection overhead to a less sensitive, higher throughput data stream.

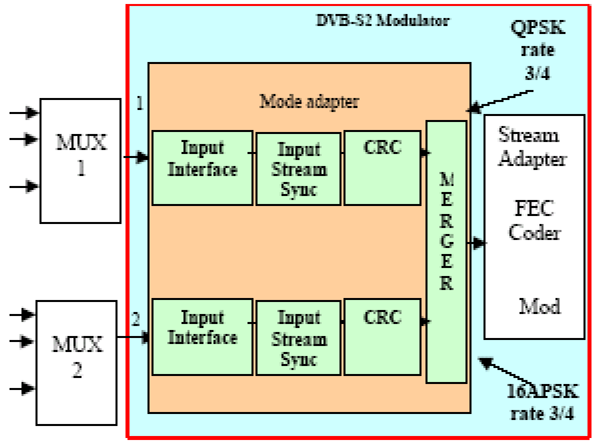

Figure J - 8 Notional VCM System Diagram

3.3. ADAPTIVE CODING AND MODULATION

In a system employing a suitable return channel, such as DVB-RCS as described in Annex I, the DVB-S2 system can accept link quality and modulation and encoding requests from deployed terminals and vary modulation and encoding of the transmitted signal on a frame-by-frame basis. ACM implementation examples are detailed in Annex H.4 of EN 302 307 and described below.

According to the negotiation between the user terminal and the ACM Routing Manager, an ACM router may separate IP packets per user, per required error protection, and per service level. The aggregate input traffic on the various protection levels shall not overload the available channel capacity, although peak input traffic may temporarily exceed channel capacity subject to input buffering capacity and service requirements on maximum delay. When total offered traffic exceeds channel capacity, lower priority packets may be delayed or dropped in deference to higher priority traffic.

ACM control loop delays should be accounted for in link margin calculations to ensure reliable operation under fast fading conditions. If control loop delays do not allow for error free reception, real-time and error sensitive services may be permanently allocated to a high protection branch. The polling strategy of the input buffers may be statically or dynamically profiled according to the traffic statistics, the propagation characteristics, and the traffic prioritization policy of the service operator.

3.3.1. Differentiated Service Levels

One method for the ACM router to interface with the DVB-S2 modulator is via multiple MPEG-2 TS or generic stream inputs, with one input per active protection level. In this case, the merger/slicer function of the DVB-S2 system partially covers the functionality of the ACM router. As shown in Figure J-9 this method resembles that of a VCM system, with dynamic encoding/modulation accomplished by the ACM router switching traffic to the appropriate stream based on return channel feedback from the remote terminal.

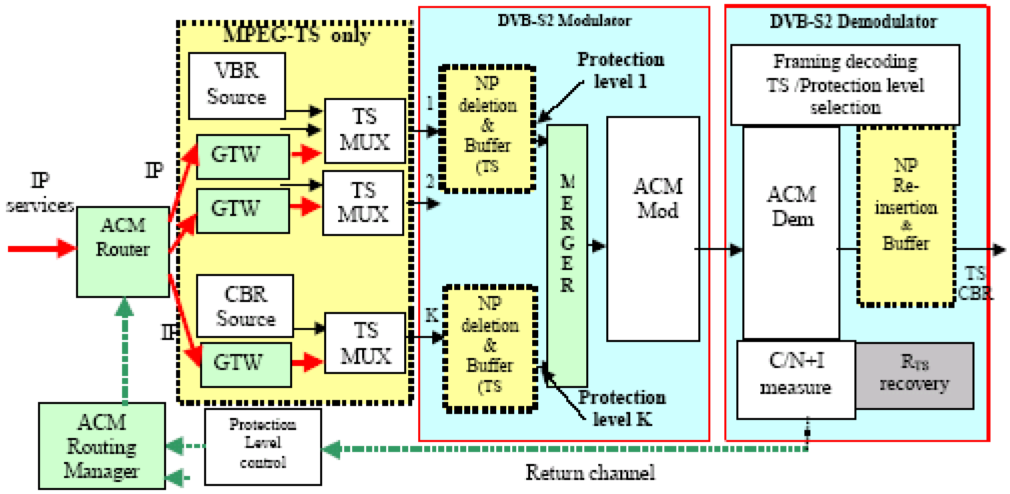

Figure J - 9 ACM System using Differentiated Service Levels

The ACM router splits user packets per service level and per required protection level and sends them to multiple DVB-S2 input interfaces, each interface permanently assigned to a given protection level. Each stream is permanently assigned a given protection level and includes the aggregate of traffic for all users assigned to that protection level. The DVB-S2 merger/slicer cyclically polls the inputs and processes the data according to the method described in this Annex. A timeout may be defined in order to prevent long delays in the absence of user data on a particular stream. Control loop delays should be minimized in order to exploit the full potential of the ACM system and prevent data and signal loss.

The dotted boxes in Figure J-9 highlight IP encapsulators, TS multiplexers, and the null packet deletion and TS buffer functions of the DVB-S2 system. These systems are only applicable to architectures employing the MPEG-2 TS input format, as described in Annex D.

3.3.2. Non-Uniform Error Protection

A second possible implementation for ACM is illustrated in Figure J-10. In this method the ACM Routing Manager performs all functions relating to traffic routing and buffering decisions and multiplexing of the traffic stream onto a single channel. The routing manager may separate packets per user, per required error protection, and per service level. The resulting IP stream (packetized generic stream) is provided to the DVB-S2 system input along with the ACM command from the user for immediate encoding and modulation according to the specified MODCOD.

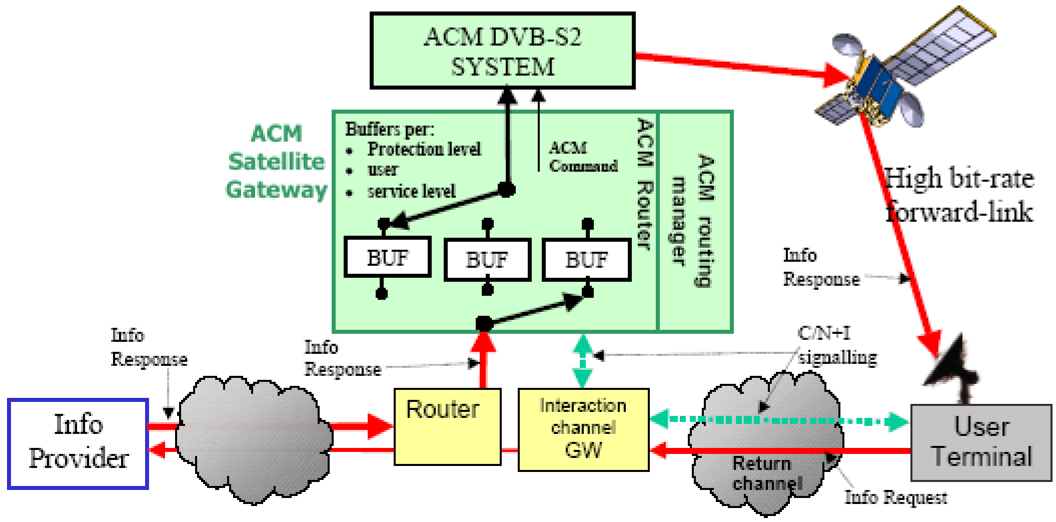

Figure J - 10 ACM System using Individual Protection Levels

Utilizing this system, the system integrator may use any routing policy. Routing and quality of service (QoS) policies should effectively handle periods where offered traffic exceeds available channel capacity, buffering or discarding packets when necessary and according to the QoS policy. Critical, time-sensitive services may be permanently directed to a high-protection branch to ensure reliable delivery.