TIA/EIA/IS-136.2-A

TIA/EIA INTERIM STANDARD

TDMA Cellular/PCS - Radio Interface - Mobile Station - Base Station Compatibility - Traffic Channels and FSK Control Channel

TIA/EIA/IS-136.2-A

(Revision of TIA/EIA/IS-136.2)

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PREFACE

These technical requirements form a compatibility standard for PCS/cellular mobile telecommunications systems. Their purpose is to ensure that a mobile station can obtain service in any PCS/cellular system manufactured according to this standard. These requirements do not address the quality or reliability of that service, nor do they cover equipment performance or measurement procedures.

To ensure compatibility (see Note 1), it is essential that both radio-system parameters and call-processing procedures be specified. The equipment and interface parameters commonly encountered in two-way radio systems have been updated and expanded to reflect the unique radio plan upon which PCS/cellular systems are based. The sequence of call processing steps that the mobile stations and base stations execute to establish calls has been specified along with the digital control messages and analog signals that are exchanged between the two stations.

The base station is subject to fewer compatibility requirements than the mobile station. Radiated power levels, both desired and undesired, are fully specified for mobile stations to control the RF interference that one mobile station can cause another. Base stations are fixed in location and their interference is controlled by proper layout and operation of the system in which the station operates. Detailed call-processing procedures are specified for mobile stations to ensure a uniform response to all base stations. Base station call procedures, like power levels, are not specified in detail because they are a part of the overall design of the individual land system. This approach to writing the compatibility specification provides the land system designer with sufficient flexibility to respond to local service needs and to account for local topography and propagation conditions.

The basic radio-system parameters and call-processing procedures for analog mode of operation embodied in the compatibility specification were originally derived from the Chicago and Baltimore-Washington developmental cellular systems and include certain additions and modifications gained by experience with the operation of commercial systems. The basic radio system parameters and call-processing procedures embodied in the dual-mode specification were derived by due process within EIA/TIA TR45.3, but have not been subject to field trial.

As commercial systems evolve there may be a need for additional capabilities primarily in the area of call-processing procedures and new system features. It is important that evolutionary changes be readily accommodated. To that end, these technical requirements have been organized into six general sections. Alterations to 2 and 3 can affect fundamental mobile station - base station compatibility. All other sections may be altered without affecting basic compatibility.

The following is a summary of each section:

1. **General.** This section comprises a list of brief explanations of terms, processes, and functions used in these requirements. Since it is the intention of these requirements to permit great latitude of system configurations and the implementation of system features, only those items required for compatibility have strict definitions. Other items may be interpreted to fit the needs of manufacturers and system operators. For example, analog control channels may be implemented with either combined paging/access functions or as separate paging and access channels. In addition, the section provides a description of the digital traffic channel structure.

2. **Mobile Station Compatibility Requirements**. This section comprises the fundamental signaling compatibility requirements of mobile stations. If strictly adhered to, a mobile station technically will be able to signal a base station. This section assures communications only if service is not otherwise restricted by operational or RF signal level constraints. For example, service may be denied for reasons of subscriber credit or because the mobile station is out of the effective range of a base station. In general, changes or alterations to this section will affect fundamental mobile station - base station compatibility and the ability of mobile stations to signal base stations irrespective of operational or RF signal level conditions.

3. **Base Station Compatibility Requirements.** This section comprises the fundamental signaling compatibility requirements of base stations and is organized in a manner similar to Section 2. (In fact, Sections 2 and 3 should be read together for a clearer understanding of the bi-directional signaling protocol.) If strictly adhered to, a base station technically will be able to signal a mobile station. As in Section 2, communications are assured only if not otherwise restricted by factors such as RF signal levels or operational limitations. In general, changes or alterations to this section will affect fundamental mobile station - base station compatibility and the ability of mobile stations to signal base stations irrespective of operational or RF signal level conditions.

4. **Requirements for Mobile Station Options.** This section states requirements for use of optional functions and features by dual-mode mobile stations. It is concerned with evolutionary changes which do not affect fundamental compatibility but which require strict definition to ensure uniform recognition and implementation of such factors as the order qualifier definitions, extended message protocols, feature coding recommendations, etc. Requirements in this section do not affect the operation of existing mobile stations. Also unaffected is the ability of mobile stations incorporating any of these options to communicate with existing base stations.

5. **Requirements for Base Station Options.** This section states requirements for use of optional functions and features by base stations. This section is in general organized to follow the sequence of items listed in 4. The reader may thus review the changes in both mobile stations and base stations by referring to corresponding paragraphs in 4 and 5. Similar to the requirements for mobile station options, this section defines changes that require strict definition to ensure uniform recognition and utilization of such factors as reserved bits, order qualifier definitions, extended message protocols, feature coding recommendations. etc. Requirements in this section do not affect the operation of existing mobile stations. Also unaffected is the ability of existing mobile stations to communicate with base stations incorporating any of these options.

6. **Change History.** This section traces all changes to these technical requirements beginning with the initial release of this standard. A brief description of each change as well as a reference to the affected section(s) are provided.

NOTES:

1. In a subscriber's home system, all call placement must be automatic. It is preferable that call placement be automatic when a mobile station roams outside of its home system.
2. The term "dual-mode mobile station" is defined as one capable of analog or digital operation.
3. This compatibility specification is based upon the specific US spectrum allocation for cellular and PCS systems.
4. Technical details are included for the operation of multiple systems in a geographic area, each with a separate set of control channels.
5. IS-137, "TDMA Cellular/PCS - Radio Interface - Minimum Performance Standards for Mobile Stations, Revision A" and IS-138, "TDMA Cellular/PCS - Radio Interface - Minimum Performance Standards for Base Stations, Revision A," provide specifications and measurement methods for cellular equipment.
6. Each system is identified by a unique 15-bit digital code, the SID code (see Section 2.3.8). The Federal Communications Commission assigns SID codes when system construction permits are issued.
7. Each mobile station is assigned a unique 32-bit binary serial number which cannot be changed by the subscriber without rendering the mobile station inoperative. (see Section 2.3.2).
8. In the message formats used between the dual-mode mobile stations and base stations, some bits are marked as reserved (RSVD). Some or all of these reserved bits may be used in the future for additional messages. Therefore, all dual-mode mobile stations and base stations must set all bits that they are programmed to treat as reserved bits to '0' (zero) in all messages that they transmit unless otherwise specified. All mobile stations and base stations must ignore the state of all bits that they are programmed to treat as reserved bits in all messages that they receive.

   In the specific case of overhead messages on the Forward Control Channel, if the mobile station receives a BCH-code-correct but unrecognizable overhead message (including Global Action Message types), the mobile station must count that message as part of the train for NAWC-counting purposes, but must not attempt to execute the message. All other messages and fields of an overhead message train that carries a message type herein indicated as 'Reserved' shall be decoded and used as appropriate.

   Implementors of mobile stations are cautioned that many other functions and features are deployed on the FOCC than those described in this standard. These functions frequently employ bits indicated herein as 'Reserved.' Reference may be made to the current version of TSB-70 for details.
9. Reserved.
10. RF Emissions. Minimum advisory standards of ANSI and the processing guidelines of FCC are contained in ANSI/IEEE C95.1-1992 Advisory Standards and FCC Rules and Regulations respectively. Members should also take notice of the more stringent exposure criteria for the general public and for radio frequency carriers with low frequency amplitude modulation as given in NCRP Report No. 86.
11. Reserved.
12. Reserved.

13. The allocation of SID numbers is under review by EIA/TIA TR45 for potential revision to accommodate international requirements. Utilization of SID numbers must be coordinated.

14. Although the analog mode of operation draws upon EIA/TIA 553, some modifications have been made.

15. All lines added or modified during the transition from IS-136 plus IS-136 Addendum to IS-136A are denoted by a single vertical change bars (|) in the right hand margin.

16. A potential EIA-553/TIA/EIA 627 compatibility problem exists as a result of differences in access channel boundary determination procedures supported in these two standards. Recommended solutions to this potential compatibility problem are as follows:

    Preferred Solution

    Section 2.3.7 (First Paging Channel) specifies two first paging channels (FIRSTCHPp-pri and FIRSTCHPp-sec) which must be stored in an TIA/EIA 627 mobile station and used to identify the first paging channel in the primary and secondary paging channel scans when the mobile station is operating in its home system. Defaulting these two values to the preferred system's (i.e., A or B band) first dedicated control channel for the primary and secondary channel sets (834.990 MHz/879.990 MHz and 835.020 MHz/880.020 MHz respectively) will prevent paging/access channels from being calculated differently when the TIA/EIA 627 mobile station operates on a EIA-553 based home system. This solution is used today and should continue to be used to ensure full interoperability of EIA-553 and TIA/EIA 627 mobile stations on both EIA-553 and TIA/EIA 627 type systems. This solution does, however, require that both home and roaming TIA/EIA 627 mobile stations use the same paging channel set (i.e., no split home-roam paging channels) as long as only a single set of dedicated controls are allocated (i.e., channels 334 - 354 for the B band).

    Non-Preferred Solution

    If a second portion of the existing spectrum is allocated for control channel use (over and above the dedicated control channels) then split home-roam paging can still be achieved for both TIA/EIA 627 and EIA-553 mobile stations. This second portion of spectrum could be managed as follows:

    - Used exclusively by home TIA/EIA 627 mobile stations, having appropriate NAM programming, for both paging and access functions or,

    - Used by home TIA/EIA 627 mobile stations, having appropriate NAM programming, for both paging and access functions and by home EIA-553 mobile stations, having appropriate NAM programming, for paging functions only. Home EIA-553 mobile stations would continue to use the existing dedicated control channels for access functions.

17. Forward control channel mobile station control messages of greater than five words in length have been shown to yield compatibility problems in some mobile stations. Implementors of systems are advised that the functions performed by these optional messages may be achieved on assigned voice channels without causing compatibility issues. Mobile Station manufacturers are advised that the length of forward control channel messages defined in future standards may be different from that defined in this standard.

18. Analog Voice Channel (AVC) and Analog Control Channel (ACC) operation is only supported in the 800 MHz hyperband.

19. The use of the global action messages Random Challenge A and Random Challenge B have been shown to yield compatibility problems in some mobile stations. Implementors of systems are advised that these problems may be reduced if these messages are not transmitted in all overhead message trains.

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1. General

1.1 Definitions

A-key. A secret, 64-bit pattern stored in the mobile station. It is used to generate/update the mobile station's Shared Secret Data. The A-key is used in the mobile station authentication process.

Abbreviated Alert. The abbreviated alert order is used to remind the user that previously selected alternative routing features are still active.

Analog Access Channel. An analog control channel used by a mobile station to access a system to obtain service.

Analog Color Code. An analog signal (see Supervisory Audio Tone) transmitted by a base station on an analog voice channel and used to detect capture of a mobile station by an interfering base station or the capture of a base station by an interfering mobile station.

Analog Control Channel (ACC). A channel used for the transmission of digital control information from a base station to a mobile station or from a mobile station to a base station.

Analog Paging Channel. A forward analog control channel that is used to page mobile stations and send orders.

Analog Voice Channel. A channel on which a voice conversation occurs and on which brief digital messages may be sent from a base station to a mobile station or from a mobile station to a base station.

AUTH. A 1-bit field in the System Parameter Overhead message. When set to 1, it signifies that the system supports the Authentication procedures.

Authentication. A procedure used by base stations to validate a mobile station's identity at system access.

Authentication Response (AUTHR). An 18-bit output of the authentication algorithm. It is used to validate mobile station registrations, originations and terminations.

Base Station. A station in the Domestic Public Cellular/PCS Radio Telecommunications Service, other than a mobile station, used for radio communications with mobile stations.

Base Station Authentication Response (AUTHBS). An 18-bit pattern generated by the authentication algorithm. AUTHBS is used to confirm the validity of base station orders to update the Shared Secret Data.

Base Station Random Variable (RANDBS). A 32-bit random number generated by the mobile station for use in authenticating base station orders to update the Shared Secret Data.

BCH Code. Bose-Chaudhuri-Hocquenghem Code.

BSMC. Base station manufacturer code (see Annex B of IS-136.1).

Busy-Idle Bits. The portion of the data stream transmitted by a base station on a forward analog control channel that is used to indicate the current Busy-Idle status of the corresponding reverse analog control channel.

Channel Quality Measurement (CQM). A digital message in two parts in which the results of mobile channel quality measurements are sent to a base station over the FACCH or the SACCH.

Coded Digital Control Channel Locator (CDL). An 11-bit data field containing the 7-bit DL and 4 protection bits.

Coded Digital Verification Color Code (CDVCC). A 12-bit data field containing the 8-bit DVCC and 4 protection bits, sent in each time slot to and from mobile stations and base stations. It is used to indicate that the correct rather than co-channel data is being decoded.

Continuous-Transmission. A mode of operation in which Discontinuous-Transmission is not permitted.

Control Mobile Attenuation Code (CMAC). A 3-bit field in the Control-Filler message that specifies the maximum authorized power level for a mobile station transmitting on a reverse control channel.

Cyclic Redundancy Check (CRC). A process in which a desired sequence of bits is encoded in a prescribed manner to enable detection and correction of bit errors. In this Standard, certain critical bit sequences are encoded using specified polynomials and procedures which use CRC-16 (ITU) and BCH code structures.

Dedicated Control Channels. A channel used for the transmission of digital control information from either a base station or a mobile station.

Digital Color Code (DCC). A digital signal transmitted by a base station on a forward analog control channel that is used to detect capture of a base station by an interfering mobile station.

Digital Control Channel Locator (DL). A 7-bit data field that is sent by the base station to help a mobile station to find a Digital Control Channel.

Digital Mobile Attenuation Code (DMAC). A 4-bit field commanding the initial mobile station power level when assigning a mobile station to a digital traffic channel.

Digital Verification Color Code (DVCC). A digital 8-bit code that is sent by the base station to the mobile station and is used for the generation of the CDVCC.

Discontinuous-Transmission (DTX). A mode of operation in which a mobile station transmitter autonomously switches between two transmitter power levels while the mobile station is in the conversation state on an analog voice channel or a digital traffic channel.

Fast Associated Control Channel (FACCH). A blank-and-burst channel used for signaling message exchange between the base station and the mobile station.

Flash Request. An indication sent on an analog voice channel from a mobile station to a base station indicating that a user desires to invoke special processing.

Flash With Info. A message sent over the digital traffic channel in either direction to indicate that special processing is required.

Forward Analog Control Channel (FOCC). An analog control channel used from a base station to a mobile station.

Forward Analog Voice Channel (FVC). An analog voice channel used from a base station to a mobile station.

Forward Digital Traffic Channel (FDTC). A digital channel from a base station to a mobile station used to transport user information and signaling. There are two separate control channels associated with the FDTC: the Fast Associated Control Channel (FACCH) and the Slow Associated Control Channel (SACCH).

Group Identification. A subset of the most significant bits of the system identification (SID) that is used to identify a group of cellular systems.

Handoff. The act of transferring a mobile station from one channel to another.

Home Mobile Station. A mobile station that operates in the cellular/PCS system from which service is subscribed.

Home System. The system which is transmitting a SID which is recognized by the mobile station as the "Home" SID.

Location Registration (LREG). A 1-bit field used to indicate the Location-Area ID Registration status.

Mean Output Power. Defined as the calorimetric power measured during the active part of transmission.

Message. There are 2 types of messages sent between base stations and mobile stations, order messages and acknowledgment messages. An order message commands or requests the recipient to take some action. In some cases, the recipient acknowledges an order message by returning an acknowledgment message. In other cases, no acknowledgment message is returned. If a message has "Ack" as part of its name, it is an Ack message; otherwise, it is an Order message. The following are examples of valid order-message names: Send Burst DTMF, Send Burst DTMF Order, Send Burst DTMF Order message, and Send Burst DTMF message. The following are examples of valid acknowledgment-message names: Send Burst DTMF Ack, Send Burst DTMF Ack message, Measurement Order Ack, and Measurement Order Ack message.

Mobile Assisted Handoff (MAHO). A process where a mobile station in digital mode, under direction from a base station, measures signal quality of specified RF channels. These measurements are forwarded to the base station upon request to assist in the handoff process.

Mobile Identification Number (MIN). The 34-bit number that is a digital representation of the 10-digit directory telephone number assigned to a mobile station.

Mobile Protocol Capability Indicator (MPCI). A 2-bit field used to indicate the mobile station's capabilities.

Mobile Station. A station in the Domestic Public Cellular/PCS Radio Telecommunications Service intended to be used while in motion or during halts at unspecified points. It is assumed that mobile stations include portable units (e.g., hand-held personal units) and units installed in vehicles.

Mobile Station Class. Mobile station classes are defined in Table 2.1.2.2.1–1 for 800 MHz operation and Table 2.1.2.2.2-1 for 1900 MHz operation.

Power Down Registration (PDREG). A 1-bit field used to indicate the Power Down Registration status.

Power Up Registration (PUREG). A 1-bit field used to indicate the Power Up Registration status.

Numeric Information. Numeric information is used to describe the operation of the mobile station. The following subscripts are used to clarify the use of the numeric information:

- "s" to indicate a value stored in a mobile station's temporary memory,
- "sv" to indicate a stored value that varies as a mobile station processes various tasks,
- "sl" to indicate the stored limits on values that vary,
- "r" to indicate a value received by a mobile station over a forward channel,
- "p" to indicate a value set in a mobile station's permanent security and identification memory, and
- "s-p" to indicate a value stored in a mobile station's semi-permanent security and identification memory.

The numeric indicators are:

- $ACCOLC_p$. A 4-bit number used to identify which overload class field controls access attempts.
- BIS_s. Identifies whether a mobile station must check for an idle-to-busy transition on a reverse analog control channel when accessing a system.
- $CCLIST_s$. The list of analog control channels to be scanned by a mobile station processing the Directed-Retry task (see Section 2.6.3.14).
- $CMAX_s$. The maximum number of channels to be scanned by a mobile station when accessing a system.
- $COUNT_{s-p}$. A modulo-64 count held in the mobile station. $COUNT_{s-p}$ is maintained during power off.
- CPA_s. Identifies whether the access functions are combined with the paging functions on the same set of analog control channels.
- DCC_s. A DCC value stored in a mobile station's temporary memory.
- DTX_s. Identifies in what way the mobile station is permitted to use the Discontinuous-Transmission mode on the analog voice channel.
- $DVCC_s$. A DVCC value stored in a mobile station's temporary memory.
- $DVCC_r$. A DVCC value received by a mobile station over the forward digital traffic channel.
- E_s. The stored value of the E field sent on the forward analog control channel. E_s identifies whether a home mobile station must send only $MIN1_p$ or both $MIN1_p$ and $MIN2_p$ when accessing the system.
- EX_p. Identifies whether home mobile stations must send $MIN1_p$ or both $MIN1_p$ and $MIN2_p$ when accessing the system. EX_p differs from E_s in that the information is stored in the mobile station's security and identification memory.
- $FIRSTCHD_s$. The number for the first channel used as a dedicated control channel.
- $FIRSTCHA_s$. The number of the first analog control channel used for accessing a system.
- $FIRSTCHP_{p-pri}$. The number of the first paging channel used as a primary paging channel in the mobile station's Home System.
- $FIRSTCHP_s$. The number of the first analog control channel used for paging mobile stations.
- $FIRSTCHP_{p-sec}$. The number of the first paging channel used as a secondary paging channel in the mobile station's Home System.
- $LASTCHA_s$. The number of the last analog control channel used for accessing a system.
- $LASTCHD_s$. The number for the last channel used as a dedicated control channel.
- $LASTCHP_s$. The number of the last analog control channel used for paging mobile stations.

- $LOCAID_{s-p}$. Identifies the current location area.
- $LOCAID_s$. The received location area identity.
- $LRCC_s$. The last registration control channel used by a mobile station.
- $LREG_s$. The stored value of the LREG field received in the most recent Location Area Global Action message.
- LT_s. Identifies whether the next access attempt is required to be the last try.
- $MIN1_p$. The 24-bit number that corresponds to the 7-digit directory telephone number assigned to a mobile station.
- $MIN2_p$. The 10-bit number that corresponds to the 3-digit area code assigned to a mobile station.
- $MAXBUSY_{sl}$. The maximum number of busy occurrences allowed on a reverse analog control channel.
- $MAXSZTR_{sl}$. The maximum number of seizure attempts allowed on a reverse analog control channel.
- N_s . The number of analog paging channels that a mobile station must scan.
- $NBUSY_{sv}$. The number of times a mobile station attempts to seize a reverse analog control channel and finds the reverse control channel busy.
- $NSZTR_{sv}$. The number of times a mobile station attempts to seize a reverse analog control channel and fails.
- $NXTREG_{s-p}$. Identifies when a mobile station must make its next registration to a system.
- PCI_s. The stored value of the PCI field in the System Parameter Overhead message.
- $PDREG_s$. The stored value of the PDREG field received in the most recent Location Area Global Action message.
- PL_s. The mobile station RF power level.
- $PUREG_s$. The stored value of the PUREG field received in the most recent Location Area Global Action message.
- $PUREG_{s-p}$. The semi-permanent value of $PUREG_s$.
- R_s. Indicates whether registration is enabled or not.
- $RAND_s$. The stored value of RAND.
- RCF_s. Identifies whether the mobile station must read a Control-Filler message before accessing a system on a reverse analog control channel.
- $REGID_s$. The stored value of the last registration number ($REGID_r$) received on a forward analog control channel.
- $REGINCR_s$. Identifies increments between registrations by a mobile station.
- S_s. Identifies whether the mobile station must send its serial number when accessing a system.

- SCC_s. A digital number that is stored and used to identify which SAT frequency a mobile station should be receiving.
- $SDCC1_s$. The SDCC value stored in a mobile station's temporary memory.
- $SDCC2_s$. The SDCC value stored in a mobile station's temporary memory.
- SID_p. The home system identification stored in the mobile station's permanent security and identification memory.
- SID_r. The system ID received on a paging or access channel.
- SID_s. The system ID received on a dedicated control channel.
- SID_{s-p}. Identifies the system of current (last successful) registration.
- $WFOM_s$. Identifies whether a mobile station must wait for an Overhead message Train before accessing a system on a reverse analog control channel.

Order. See definition for message.

Overload Control (OLC). A means to restrict reverse control channel accesses by mobile stations. Mobile stations are assigned one (or more) of sixteen control levels. Access is selectively restricted by a base station setting one or more OLC bits in the Overload Control Global Action message.

Paging. The act of seeking a mobile station when an incoming call has been placed to it.

Personal Identification Number (PIN). A secret number managed by the system operator for each subscriber. The PIN is intended primarily for use in authenticating the subscriber.

Physical Layer Control. A digital-mode-base station control message to initiate or change certain mobile station parameters such as traffic channel power, time alignment, and whether Discontinuous-Transmission (DTX) is permitted.

Primary Paging Channels. A forward analog control channel that is used to page mobile stations and send orders, and is supported by both EIA-553 and IS-54 compatible mobile stations.

Privacy Mode (PM). A 1-bit parameter used to refer to the Voice Privacy status: 0 = off, 1 = on.

Protocol Capability Indicator (PCI). A 1-bit field in the first word of the System Parameter Overhead message that when set to one indicates the base station is capable of digital operation.

Protocol Version (PV). A 4-bit field used to indicate the mobile station or a base station capabilities.

Protocol Version Indicator (PVI). A 1-bit field used to indicate whether a base station is TIA/EIA 627 capable or IS-136 capable.

Random Variable (RAND). A 32-bit random number issued periodically by the base station in two 16-bit pieces: RAND1_A and RAND1_B. The mobile station stores and uses the most recent version of RAND in the authentication process.

Random Variable Confirmation (RANDC). A 8-bit number used to confirm the last RAND received by the mobile station.

Registration. The steps by which a mobile station identifies itself to a base station as being active in the system at the time the message is sent to the base station.

Release Request. A message sent from a mobile station to a base station indicating that the user desires to disconnect the call.

Reverse Analog Control Channel (RECC). The analog control channel used from a mobile station to a base station.

Reverse Analog Voice Channel (RVC). The analog voice channel used from a mobile station to a base station.

Reverse Digital Traffic Channel (RDTC). A digital channel from a mobile station to a base station used to transport user information and signaling. There are two separate control channels associated with the RDTC: The Fast Associated Control Channel (FACCH) and the Slow Associated Control Channel (SACCH.)

Roamer. A mobile station that operates in a cellular system other than the one from which service is subscribed.

Scan of Channels. The procedure by which a mobile station examines the signal strength of each forward analog control channel.

Secondary Control Channels. A supplementary set of analog control channels developed specifically for IS-54 compatible mobile stations. Such channels are used for the transmission of digital control information from either the base or mobile stations.

Secondary Paging Channels. In addition to the primary paging channels, a supplementary set of analog control channels developed specifically for IS-54 compatible mobile stations. Such channels are used to page mobile stations and send orders.

Seizure Precursor. The initial digital sequence transmitted by a mobile station to a base station on a reverse analog control channel.

Shared Secret Data (SSD). A 128-bit pattern stored in the mobile station (in semi-permanent memory) and known by the base station. SSD is a concatenation of two 64-bit subsets: SSD-A, which is used to support the Authentication procedures, and SSD-B, which serves as one of the inputs to the voice privacy mask generation process. Shared Secret Data is maintained during power off.

Shared Secret Data Random Variable (RANDSSD). A 56-bit random number generated by the mobile station's home system. RANDSSD is used in conjunction with the mobile station's A-key and ESN to generate its Shared Secret Data.

Short Message Services (SMS). Service used to transfer short messages to or from a mobile station.

Signaling Tone. A 10-kHz tone transmitted by a mobile station on an analog voice channel to: 1) confirm orders, 2) signal flash requests, and 3) signal release requests.

Slow Associated Control Channel (SACCH). A continuous channel used for signaling message exchange between the base station and the mobile station. A fixed number of bits are allocated to the SACCH in each TDMA slot.

SOC. System Operator Code (see Section 6.5 and Annex B of IS-136.1).

SOR. Standard Offset Reference (see Section 1.2.1.1).

Status Information. The following status information is used in this section to describe mobile station operation:

- Serving-System Status. Indicates whether a mobile station is tuned to channels associated with System A or System B.
- First-Registration ID Status. A status variable used by the mobile station in association with its processing of received Registration ID messages.
- First-Location-Area ID Status. A status variable used by the mobile station in association with its processing of received Location Area ID messages.
- Location-Registration ID Status. A status variable used by the mobile station in association with its processing of Power Up Registrations and location-based registrations.
- First-Idle ID Status. A status variable used by the mobile station in association with its processing of the Idle task.
- Local Control Status. Indicates whether a mobile station must respond to Local Control messages.
- Roam Status. Indicates whether a mobile station is in its home system.
- Termination Status. Indicates whether a mobile station must terminate the call when it is on an analog voice channel.
- SOC Signaling Status. Indicates whether or not mobile station SOC signaling is currently enabled for the mobile station.
- BSMC Signaling Status. Indicates whether or not mobile station BSMC signaling is currently enabled for the mobile station.
- Base Station Protocol Version Status. Indicates whether or not the base station supports IS-136 protocol.

STU-III. U.S. government's Secure Terminal Unit that provides end-to-end secure voice and data services over cellular and landline networks.

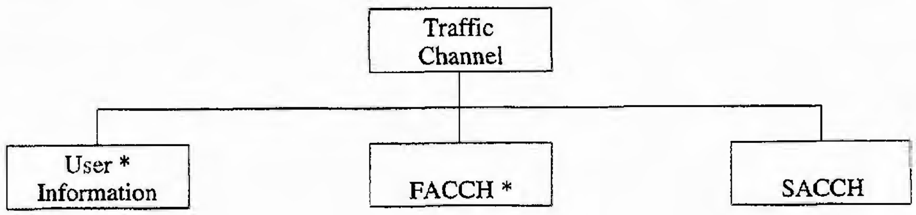
Supervisory Audio Tone (SAT). One of three tones in the 6-kHz region that are transmitted by a base station and transponded by a mobile station.

Supplementary Digital Color Code (SDCC1, SDCC2). Additional bits assigned to increase the number of color codes from four to sixty-four, transmitted on the forward analog control channel.

Symbol. In the $\pi/4$ DQPSK modulation scheme specified, each symbol carries 2 bits of information.

System Identification (SID). A digital identification associated with a cellular system; each system is assigned a unique number.

Traffic Channel. That portion of the digital information transmitted between the base station and the mobile station, or between the mobile station and the base station, that is dedicated to the transport of user and signaling information as depicted in the figure below.



* FACCH and user information cannot be sent simultaneously.

Unique Challenge Authentication Response (AUTHU). An 18-bit pattern generated by the authentication algorithm. AUTHU is used to support the Unique Challenge-Response procedure.

Unique Challenge-Response Procedure. An exchange of information between a mobile station and a base station for the purpose of confirming the mobile station's identity. The procedure is initiated by the base station and is characterized by the use of a challenge-specific random number (i.e., RANDU) instead of the random variable broadcast globally (RAND).

Unique Random Variable (RANDU). A 24-bit random number generated by the base station in support of the Unique Challenge-Response procedure.

User Alert. The User Alert order is sent to activate user alerting at a mobile station.

Voice Mobile Attenuation Code (VMAC). A 3-bit field in the Extended Address Word commanding the initial mobile station power level, when assigning a mobile station to an analog voice or traffic channel.

Voice Privacy. The process by which user voice transmitted over a digital traffic channel is afforded a modest degree of cryptographic protection against eavesdropping in the mobile station-base station segment of the connection.

Wait-for-Overhead Message (WFOM). A 1-bit field in the Control-Filler message that when set to one causes the mobile station to wait for an overhead message before transmitting on a reverse control channel.

1.2 Digital Traffic Channel Structure

This diagram depicts the frame structure:

Figure 1.2–1 TDMA Frame Format

One Frame = 1944 bits (972 Symbols) = 40 ms. (25 frames per second)

Slot 1	Slot 2	Slot 3	Slot 4	Slot 5	Slot 6
				←→ One Slot	

Figure 1.2–2 Time Slot Formats

6	6	16	28	122	12	12	122
G	R	DATA	SYNC	DATA	SACCH	CDVCC	DATA

Slot Format Mobile Station to Base Station (All numbers indicate bits)

28	12	130	12	130	1	11
SYNC	SACCH	DATA	CDVCC	DATA	RSVD = 1	CDL

Slot Format Base Station to Mobile Station (All numbers indicate bits)

The Bit Positions (BP) of forward and reverse time slots are numbered sequentially from 1 to 324.

In the forward time slot, the first transmitted bit of the SYNC field has BP = 1 and the last transmitted bit of the CDL field has BP = 324. In the reverse time slot, the first transmitted bit of the G field has BP = 1.

Interpretation of the data fields is as follows:

- G – Guard Time (see Section 1.2.3)
- R – Ramp Time (see Section 1.2.3)
- DATA – User Information or FACCH
- SACCH – Slow Associated Control Channel (see Sections 2.7.3.1.2 and 3.7.3.1.2)
- CDVCC – Coded Digital Verification Color Code (see Sections 1.2.5, 2.4.3, and 3.4.3)
- SYNC – Synchronization and Training (see Section 1.2.4)
- CDL - Coded Digital Control Channel Locator (see Section 1.2.6)
- RSVD – Reserved

1.2.1 Frame Length

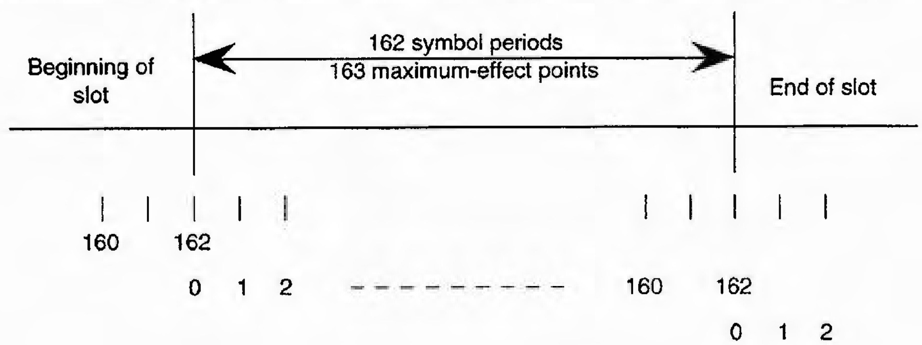
The frame length on each digital TDMA RF channel shall be 40 milliseconds. Each frame shall consist of six equally sized time slots (1–6), exactly 162 symbols in length. Each full-rate traffic channel shall utilize two equally spaced time slots of the frame (1&4, 2&5, or 3&6). Each half-rate traffic channel shall utilize one time slot of the frame.

At the mobile station, the offset between the reverse and forward frame timing, with no time advance applied, is one time slot plus 45 symbols (207 symbol periods). Time slot 1 of frame N in the forward direction occurs 207 symbol periods after time slot 1 of frame N in the reverse direction, with no time advance.

The relation between data modulation and the above definition of time slots is as follows:

> The modulation timing within a forward time slot shall be such that the first modulated symbol to be used by the mobile station receiving that time slot (the first symbol of the sync word) shall have maximum effect on the signal transmitted from the base station antenna coincident with the beginning of the time slot as shown in Figure 1.2.1–1.

Figure 1.2.1–1 Forward Time Slot Symbols

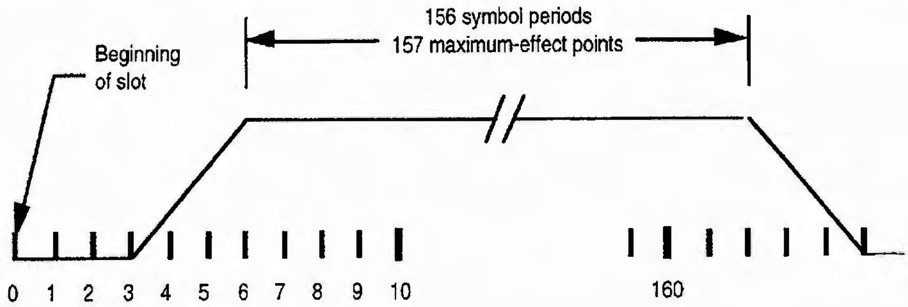


> The maximum effect point is defined to be the point in time at which the pulse shaping function associated with the modulation is at a maximum for the symbol of interest (refer to Section 2.1.3.3.1 for a description of the pulse shaping function). The relation between the maximum effect point and the associated symbol period is such that the maximum effect point is defined to occur at the end of the symbol period.

> The first maximum effect point for the forward channel is at point 0. Decoding the phase change from point 0 to point 1 provides the first two bits of data. Decoding the phase change from point 161 to point 162 provides the last two bits of data. Point 0 is the same maximum effect point as point 162 of the previous time slot.

> The modulation timing within a reverse time slot shall be such that the first modulated symbol has maximum effect on the signal transmitted at the antenna by the mobile station 6.0 symbol periods after the beginning of the reverse time slot, as shown in Figure 1.2.1–2.

Figure 1.2.1–2 Reverse Time Slot Symbols

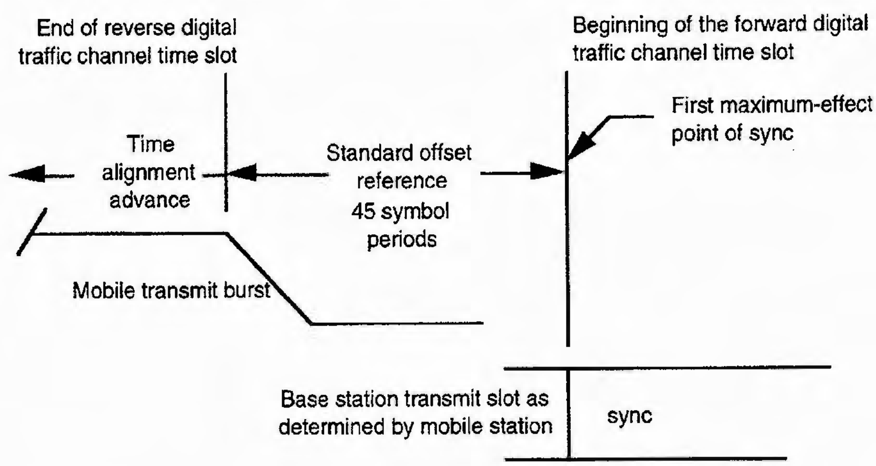


The first maximum-effect for the reverse channel is at point 6. Decoding the phase change from point 6 to point 7 provides the first two bits of data.

1.2.1.1 Standard Offset Reference

Figure 1.2.1.1–1 depicts the relationship between transmit and receive as specified for the mobile station (full-rate).

Figure 1.2.1.1–1 Standard Offset Reference



1.2.2 Gross Rate for the Traffic Channel

The gross data rate for a full-rate digital traffic channel shall be 13.0 kbit/s.

1.2.3 Guard and Power Ramp Up Interval

The interval of each (i.e., Guard & Power Ramp Up) is 3 symbols in duration.

1.2.4 Synchronization Word/Time Slot Identifier

The synchronization word/time slot identifier is a 14-symbol field which is used for slot synchronization, equalizer training, and time slot identification. For its location refer to Figure 1.2–2. Six unique synchronization sequences are defined.

The synchronization word has good autocorrelation properties to facilitate synchronization and training.

Six time slot identifiers are defined, which have good cross correlation properties. The actual synchronization sequences are defined in Table 1.2.4–2.

Line 8 from Table 1.2.4–1 identifies the sync word for a channel fully assigned to full-rate users.

Line 1 from Table 1.2.4–1 identifies the sync word for a channel fully assigned to half-rate users.

Lines 2 through 7 identify the sync word order for a mixture of full-rate and half-rate users, such that only one sync word is assigned per user.

Unassigned slots are indicated by the base station as half-rate user slots in the Time Slot Identifier field.

The mobile station uses its assigned sync word on the RDTC.

Table 1.2.4-1 Sync Word Usage

Time Slot / Line	1	2	3	4	5	6
1	1	2	3	4	5	6
2	1	2	3	1	5	6
3	1	2	3	4	2	6
4	1	2	3	4	5	3
5	1	2	3	1	2	6
6	1	2	3	1	5	3
7	1	2	3	4	2	3
8	1	2	3	1	2	3

The preferred assignment by base station of sync identity is from the table above. However, the base station, upon appropriate signaling, may assign other sync word ordering.

The preferred assignment shall be used for a particular traffic channel unless the mobile station supports extended modulation and framing in accordance with Table 3.7.1–1.

Table 1.2.4-2 Synchronization Sequences

The sync words are specified by the following **phase changes** in radians:

Sync 1	$-\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-\frac{\pi}{4}$
Sync 2	$-\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$-\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-\frac{\pi}{4}$
Sync 3	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-\frac{\pi}{4}$	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$	$\frac{\pi}{4}$	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$
Sync 4	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$
Sync 5	$\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-\frac{\pi}{4}$	$\frac{\pi}{4}$	$-\frac{\pi}{4}$	$\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$3\frac{\pi}{4}$	$\frac{\pi}{4}$	$3\frac{\pi}{4}$
Sync 6	$-3\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$	$\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$-\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$\frac{\pi}{4}$	$\frac{\pi}{4}$	$3\frac{\pi}{4}$	$-3\frac{\pi}{4}$	$3\frac{\pi}{4}$

Note: The current specification is based on the best available information at the time of writing. As such, it is subject to revision.

1.2.5 Coded Digital Verification Color Code (CDVCC)

This is a 12-bit field, permitting 255 distinct values of CDVCC. The same CDVCC may be used for all base station and mobile station transmissions in the same cell (or sector).

DVCC is an 8-bit word which is coded using a (15,11) Hamming code shortened to a (12,8) code to form a 12-bit Coded Digital Verification Color Code (CDVCC). The following coding procedure defines the relationship between the 8-bit DVCC and the 12-bit CDVCC, and does not imply an implementation. Bit d7 is the most significant bit of DVCC and is received earliest in time in the mobile station control message.

DVCC (8-bits) = (d7,d6,d5,d4,d3,d2,d1,d0)

1. Form DVCC Information Word polynomial a(X):

$$d7X^7 + d6X^6 + d5X^5 + d4X^4 + d3X^3 + d2X^2 + d1X^1 + d0X^0$$

2. Multiply a(X) by X^4.

3. Obtain the remainder b(X) from dividing $X^4a(X)$ by $X^4 + X + 1$.

$$\text{Where } \frac{a(X)X^4}{X^4 + X + 1} = q(X) + \frac{b(X)}{X^4 + X + 1}$$

4. CDVCC (12-bits) is defined as (d7,d6,d5,d4,d3,d2,d1,d0,b3,b2,b1,b0).

   CDVCC bit d7 is transmitted first.

1.2.6 Coded Digital Control Channel Locator (CDL)

This field contains a coded version of Digital Control Channel Location (DL) values, and provides information that may be used by the mobile station to assist in the location of a Digital Control Channel.

- DL to Channel Number Mapping - Cellular Frequencies
  - A properly decoded DL value indicates that a digital control channel may be found on RF channel number in the range ((8 * DL)+1) to ((8 * DL)+8), provided the RF channel number is valid (see Section 2.1.1.1).
- DL to Channel Number Mapping - PCS Frequencies
  - ((16 * DL + 1) to (16 * DL) + 16)

The CDL value zero is reserved and undefined, and therefore does not provide any digital control channel location information. A value of zero shall not be interpreted as indicating that no digital control channel is available in that cell.

The channel encoding of the DL into CDL is similar to how CDVCC is handled (see Section 1.2.5). The d_7 bit is omitted (set to zero) in the encoding process and not transmitted as part of CDL. The LSB of DL is d_0. After encoding, the check bits b_3, b_2 b_1 and b_0 are all inverted before forming the resulting CDL information. The bit positions of the CDL on the DTC are as follows:

d_6	d_5	d_4	d_3	d_2	d_1	d_0	$\bar{b}_3$	$\bar{b}_2$	$\bar{b}_1$	$\bar{b}_0$
BP_{314}	BP_{315}	BP_{316}	BP_{317}	BP_{318}	BP_{319}	BP_{320}	BP_{321}	BP_{322}	BP_{323}	BP_{324}

1.3 Timing Tolerances

Unless otherwise specified, all call-processing timers and call-processing timing values have a tolerance of ±10%. Tolerances of other parameters are provided for guidance only. Refer to IS-137, "TDMA Cellular/PCS - Radio Interface - Minimum Performance Standards for Mobile Stations, Revision A", and IS-138, "TDMA Cellular/PCS - Radio Interface - Minimum Performance Standards for Base Stations, Revision A", for minimum standards, definitions, tolerances, and measurement methods.

2. Mobile Station

(Also see Section 4 for Mobile Station Options.)

2.1 Transmitter

2.1.1 Frequency Parameters

2.1.1.1 Channel Spacing and Designation

2.1.1.1.1 800 MHz Operation

Channel spacing shall be 30 kHz and the dual-mode mobile station transmit channel at 825.030 MHz (and the corresponding base station transmit channel at 870.030 MHz) shall be termed channel number 1. The 20 MHz range of channels 1 through 666 as shown in Table 2.1.1.1.1-1 for System A and System B is basic. The additional 5 MHz of channels 667 through 799 and (wrap-around) 991 through 1023 for extending System A (A', A") and B (B') is mandatory. The station class mark (SCM, see Section 2.3.3) shall be set appropriately.

Table 2.1.1.1.1-1 Channel Numbers and Frequencies

System	Bandwidth (MHz)	Number of Channels	Boundary Channel Number	Transmitter Center Frequency (MHz)	
				Mobile	Base
(Not used)		1	(990)	(824.010)	(869.010)
A"	1	33	991	824.040	869.040
			1023	825. 000	870. 000
A	10	333	1	825.030	870.030
			333	834.990	879.990
B	10	333	334	835.020	880.020
			666	844.980	889.980
A'	1.5	50	667	845.010	890.010
			716	846.480	891.480
B'	2.5	83	717	846.510	891.510
			799	848.970	893.970

In the above, the center frequency in MHz corresponding to the channel number (expressed as N) is calculated as follows.

Transmitter	Channel Number	Center Frequency (MHz)
Mobile	$1 \le N \le 799$	$0.030\ N + 825.000$
	$990 \le N \le 1023$	$0.030\ (N - 1023) + 825.000$
Base	$1 \le N \le 799$	$0.030\ N + 870.000$
	$990 \le N \le 1023$	$0.030\ (N - 1023) + 870.000$

2.1.1.1.2 1900 MHz Operation

Channel spacing shall be 30 kHz with the mobile station and the corresponding base station transmit channels as listed in Table 2.1.1.1.2-1.

Table 2.1.1.1.2-1 Channel Numbers and Frequencies for 1900 MHz Operation

Band	Bandwidth (MHz)	Number of Channels	Boundary Channel Number	Transmitter Center Frequency (MHz)	
				Mobile	Base
Not Used		1	1	1850.010	1930.050
A	15	497	2	1850.040	1930.080
			498	1864.920	1944.960
A,D (Note 1)		1	499	1864.950	1944.990
A,D (Note 1)		1	500	1864.980	1945.020
A,D (Note 1)		1	501	1865.010	1945.050
D	5	164	502	1865.040	1945.080
			665	1869.930	1949.970
D,B (Note 1)		1	666	1869.960	1950.000
D,B (Note 1)		1	667	1869.990	1950.030
B	15	498	668	1870.020	1950.060
			1165	1884.930	1964.970
B,E (Note 1)		1	1166	1884.960	1965.000
B,E (Note 1)		1	1167	1884.990	1965.030
E	5	165	1168	1885.020	1965.060
			1332	1889.940	1969.980
E,F (Note 1)		1	1333	1889.970	1970.010
E,F (Note 1)		1	1334	1890.000	1970.040
F	5	164	1335	1890.030	1970.070
			1498	1894.920	1974.960
F,C (Note 1)		1	1499	1894.950	1974.990
F,C (Note 1)		1	1500	1894.980	1975.020
F,C (Note 1)		1	1501	1895.010	1975.050
C	15	497	1502	1895.040	1975.080
			1998	1909.920	1989.960
Not Used		1	1999	1909.950	1989.990

Note 1: This channel does not entirely fall into a single band (A,B,C,D,E or F). A mobile station capable of operating in any band (A,B,C,D,E or F or any combination of these) shall be able to operate also on the associated border channel(s).

In the above, the transmitter center frequency in MHz corresponding to the channel number (expressed as N) is calculated as follows:

Transmitter	Channel Number	Center Frequency (MHz)
Mobile	1 ≤ N ≤1999	0.030 N + 1849.980
Base	1 ≤ N ≤1999	0.030 N + 1930.020

2.1.1.2 Frequency Tolerance

2.1.1.2.1 Frequency Tolerance for Analog Mode Operation

The dual-mode mobile station carrier frequency must be maintained within ±2.5 parts per million (ppm) of any assigned channel frequency, except during channel switching (see Section 2.1.2.1). This tolerance must be maintained over the ambient temperature range of –30°C to +60°C, and over the supply voltage range of ±15 percent from the nominal value.

2.1.1.2.2 Frequency Tolerance for 800 MHz Digital Mode Operation

The dual-mode mobile station transmit carrier frequency must track within ±200 Hz of a frequency value 45.0 MHz lower than the frequency of the corresponding base station transmit signal, as measured at the mobile station receiver, except during channel switching or MAHO channel scanning (see Section 2.4.5). This tolerance must be maintained over the ambient temperature range of –30°C to +60°C and over the supply voltage range of ±15 percent from the nominal value.

2.1.1.2.3 Frequency Tolerance for 1900 MHz Digital Mode Operation

The mobile station transmit carrier frequency must track within ±200 Hz of a frequency value 80.04 MHz lower than the frequency of the corresponding base station transmit signal, as measured at the mobile station receiver, except during channel switching or MAHO channel scanning (see Section 2.4). This tolerance must be maintained over the ambient temperature range of –30°C to +60°C and over the supply voltage range of ±15 percent from the nominal value.

2.1.2 Power Output Characteristics

2.1.2.1 Carrier On/Off Conditions

2.1.2.1.1 Constant Envelope Conditions

The carrier-off condition is defined as a power output at the transmitting antenna connector not exceeding –60 dBm. When commanded to the carrier-on condition on a reverse control channel, a mobile station transmitter must come to within 3 dB of the specified output power (see Section 2.1.1.2.1) and to within the required stability (see Section 2.1.1.2) within 2 ms. Conversely, when commanded to the carrier-off condition, the transmit power must fall to a level not exceeding –60 dBm within 2 ms. Whenever a transmitter is more than 1 kHz from its initial or final value during channel switching, the transmitter carrier must be inhibited to a power output level not greater than –60 dBm.

2.1.2.1.2 $\frac{\pi}{4}$ Shifted DQPSK Conditions

The carrier-off condition is defined as a power output at the transmitting antenna connector not exceeding –60 dBm. The steady-state carrier-on condition is defined in Section 2.1.2.2. The acceptable instantaneous variation in power output level is defined in Section 2.1.3.3.1.1.3. The first 3 symbol periods of the TDMA burst are assigned to guard time (see Section 1.2). During the 3 symbol periods of guard time the carrier remains in the carrier-off condition. Symbol periods 4 through 6 are assigned to the power ramp time. The carrier-on command for the reverse traffic channel occurs at the beginning of symbol period 4. By the end of symbol period 6 the transmit power must be sufficiently stable to permit the conditions of Section 2.1.3.3.1.1.3 to be met. The beginning time of symbol period 1 is established by the Time Alignment procedure described in Section 2.1.3.3.5. Conversely, when commanded to the carrier-off condition the mobile station transmit power must fall to a level not exceeding –60 dBm within 3 symbol periods. The spectrum profile of the ramp up and ramp down must conform to the adjacent and alternate channel emission requirements given in Section 2.1.4.1.2. During the information bearing portion of the message the transmitter frequency accuracy must be within the required stability (see Section 2.1.1.2.2).

2.1.2.2 Power Output and Power Control

See IS-137-A for appropriate performance values.

2.1.2.2.1 800 MHz Operation

The mean effective radiated power (ERP) with respect to a half wave dipole for any class mobile station transmitter shall not exceed 8 dBW (6.3 Watts). An inoperative antenna assembly must not degrade the spurious emission levels as defined in Section 2.1.4.2. The nominal ERP (average burst power in digital mode) for each class of mobile station transmitter is: Class I 6 dBW (4.0 Watts), Class II 2 dBW (1.6 Watts), Class III –2 dBW (0.6 Watts), and Class IV –2 dBW (0.6 Watts). Class V, Class VI, Class VII, and Class VIII are reserved for future definition. Class IV is available only in dual-mode mobile stations.

All mobile station transmitters must be capable of reducing or increasing power on command from a base station specifying the power level 0 to 7. Mobile stations in classes IV through VIII must further be able to change power to levels in the range of power levels 0 to 10 by a Physical Layer Control message (Power Change) from the base station. Only mobile stations operating in digital mode can operate below power level 7 (see Sections 2.7 and 3.7). The nominal levels are given in Table 2.1.2.2.1–1.

The power levels 0 to 7 must be maintained within the range of +2 dB/–4 dB of its nominal level over the ambient temperature range of –30°C to +60°C, and over the supply voltage range of ±10 percent from the nominal value, accumulative. A power change command will raise or lower power in increments of 4 dB. (See Table 2.1.2.2.1–1 for tolerances.)

For power levels 8 through 10, RF power emission must be maintained within the range +2 dB/–6 dB of the initial power level unless a Physical Layer Control message (Power Change) is received, over the same temperature and supply voltage conditions stated above. A commanded increase of the power level number (PL) must never result in an increase of output power.

All classes of mobile stations will respond to a CMAC, DMAC or a VMAC command by setting their transmit power to the appropriate Mobile Station Power Level, regardless of prior Mobile Station Power Level. Mobile station power levels 8 through 10 are not allowed on analog signaling or voice channels.

Table 2.1.2.2.1–1 Mobile Station Nominal Power Levels

Mobile Station Power Level (PL)	Mobile Attenuation Code (MAC)	Nominal ERP (dBW) for Mobile Station Power Class							
		I	II	III	IV	V	VI	VII	VIII
0	0000	6	2	−2	−2	•	•	•	•
1	0001	2	2	−2	−2	•	•	•	•
2	0010	−2	−2	−2	−2	•	•	•	•
3	0011	−6	−6	−6	−6	•	•	•	•
4	0100	−10	−10	−10	−10	•	•	•	•
5	0101	−14	−14	−14	−14	•	•	•	•
6	0110	−18	−18	−18	−18	•	•	•	•
7	0111	−22	−22	−22	−22	•	•	•	•
DUAL-MODE ONLY									
8	1000	−22	−22	−22	−26 ± 3 dB	•	•	•	•
9	1001	−22	−22	−22	−30 ± 6 dB	•	•	•	•
10	1010	−22	−22	−22	−34 ± 9 dB	•	•	•	•

Note: The three least significant bits of MAC are used in the CMAC/VMAC field. All four bits of MAC are used in the DMAC field.

2.1.2.2.2 1900 MHz Operation

The mean effective radiated power (ERP) with respect to a half wave dipole for any class mobile station transmitter shall not exceed 8 dBW (6.3 Watts). An inoperative antenna assembly must not degrade the spurious emission levels as defined in Section 2.1.4.2. The nominal ERP (average burst power in digital mode) for each class of mobile station transmitter is: Class II 0 dBW (1.0 Watts), and Class IV – 2 dBW (0.6 Watts). Class III, Class V, Class VI, Class VII, and Class VIII are reserved for future definition.

All mobile station transmitters must be capable of reducing or increasing power on command from a base station specifying the power level 0 to 10. The nominal levels are given in Table 2.1.2.2.2–1.

The power levels 0 to 7 must be maintained within the range of +2 dB/–4 dB of its nominal level over the ambient temperature range of –30°C to +60°C, and over the supply voltage range of ±10 percent from the nominal value, accumulative. A power change command will raise or lower power in increments of 4 dB. (See Table 2.1.2.2.2–1 for tolerances.)

For power levels 8 through 10, RF power emission must be maintained within the range +2 dB /–6 dB of the initial power level unless a Physical Layer Control message (Power Change) is received, over the same temperature and supply voltage conditions stated above. A commanded increase of the power level number (PL) must never result in an increase of output power.

All classes of mobile stations will respond to a DMAC command by setting their transmit power to the appropriate Mobile Station Power Level, regardless of prior Mobile Station Power Level.

Table 2.1.2.2.2-1 Mobile Station Nominal Power Levels

Mobile Station Power Level (PL)	Mobile Attenuation Code (DMAC)	Nominal ERP (dBW) for Mobile Station Power Class						
		II	III	IV	V	VI	VII	VIII
0	0000	0	•	–2	•	•	•	•
1	0001	0	•	–2	•	•	•	•
2	0010	– 2	•	–2	•	•	•	•
3	0011	– 6	•	–6	•	•	•	•
4	0100	–10	•	–10	•	•	•	•
5	0101	–14	•	–14	•	•	•	•
6	0110	–18	•	–18	•	•	•	•
7	0111	–22	•	–22	•	•	•	•
8	1000	–26 ± 3 dB	•	–26 ± 3 dB	•	•	•	•
9	1001	–30 ± 6 dB	•	–30 ± 6 dB	•	•	•	•
10	1010	–34 ± 9 dB	•	–34 ± 9 dB	•	•	•	•

2.1.3 Modulation Characteristics

2.1.3.1 Analog Voice Signals

The modulator is preceded by the following five voice-processing stages (in the order listed):

- Transmit Audio Level Adjustment
- Compressor
- Pre-Emphasis
- Deviation Limiter
- Post Deviation-Limiter Filter

Pending the generation of a complete speech transmission plan for dual-mode cellular systems, the following requirements shall be met to ensure compatibility with the transmission plan for fixed digital speech networks.

Transmit Level Adjustment of Alternate Codecs

See IS-641.

Transmit Level Adjustment of the VSELP Codec

The transmit audio sensitivity shall be adjusted such that the same reference input level used to generate a state of $R_0 = 21$ in the VSELP codec results in a ±2.9 kHz peak frequency deviation of the transmitted carrier measured with a 1 kHz sinusoidal tone.

2.1.3.1.1 Compressor

This stage is the compressor portion of a 2:1 syllabic compandor. For every 2 dB change in input level to a 2:1 compressor within its operating range, the change in output level is a nominal 1 dB. The compressor must have a nominal attack time of 3 ms and a nominal recovery time of 13.5 ms as defined by the ITU (Reference: Recommendation G162, CCITT Plenary Assembly, Geneva, May-June 1964, Blue Book, Vol. 111, P. 52). The nominal reference input level to the compressor is that corresponding to a 1000 Hz acoustic tone at the expected nominal speech volume level (IS-137). This level must produce a nominal ±2.9 kHz peak frequency deviation of the transmitted carrier.

2.1.3.1.2 Pre-Emphasis

The pre-emphasis characteristic must have a nominal +6 dB/octave response between 300 and 3000 Hz.

2.1.3.1.3 Deviation Limiter

For audio (voice) inputs applied to the transmitter voice-signal processing stages, a dual-mode mobile station operating in analog mode must limit the instantaneous frequency deviation to ±12 kHz. This requirement excludes supervision signals (see Section 2.4) and wideband data signals (see Section 2.1.3.2).

2.1.3.1.4 Post Deviation-Limiter Filter

The deviation limiter must be followed by a low-pass filter whose characteristics are:

Frequency Band	Attenuation Relative to 1000 Hz
3000 – 5900 Hz	≥ 40 log (f/3000) dB
5900 – 6100 Hz	≥ 35 dB
6100 – 15000 Hz	≥ 40 log (f/3000) dB
above 15000 Hz	≥ 28 dB

2.1.3.2 Wideband Analog Data Signals

2.1.3.2.1 Encoding

The reverse control channel (RECC) and reverse voice channel (RVC) wideband data streams (see Section 2.7) must be further encoded such that each non-return-to-zero binary one is transformed to a zero-to-one transition, and each non-return-to-zero binary zero is transformed to a one-to-zero transition.

2.1.3.2.2 Modulation and Polarity

The filtered wideband data stream must then be used to modulate the transmitter carrier using direct binary frequency shift keying. A one (i.e., high state) into the modulator must correspond to a nominal peak frequency deviation 8 kHz above the carrier frequency, and a zero into the modulator must correspond to a nominal peak frequency deviation 8 kHz below the carrier frequency.

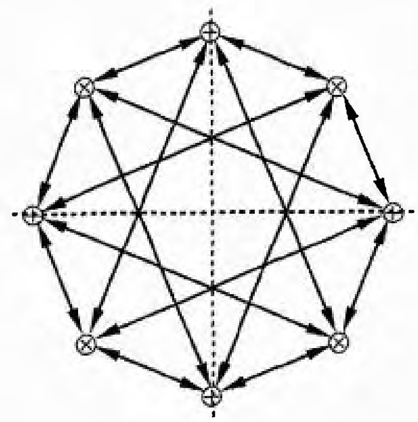
2.1.3.3 Digital Voice and Data Signals

2.1.3.3.1 Modulation

The modulation method used is known as $\pi/4$ shifted, differentially encoded quadrature phase shift keying.

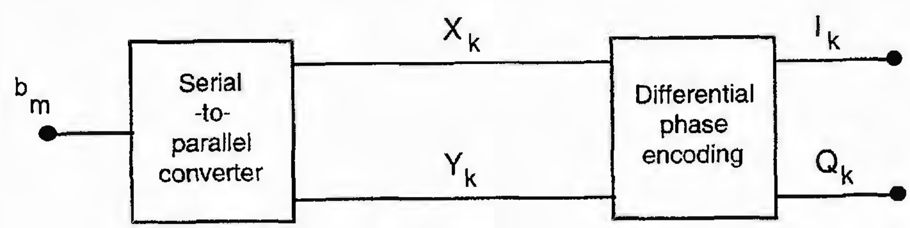
The modulation scheme uses the phase constellation shown in Figure 2.1.3.3.1-1. Note that Gray code is used in the mapping; two di-bit symbols corresponding to adjacent signal phases differ only in a single bit. Since most probable errors due to noise result in the erroneous selection of an adjacent phase, most di-bit symbol errors contain only a single bit error. Note also, the rotation by $\pi/4$ of the basic QPSK constellation for odd (denoted $\oplus$) and even (denoted $\otimes$) symbols.

Figure 2.1.3.3.1-1 Phase Constellation



The information is differentially encoded; symbols are transmitted as changes in phase rather than absolute phases. A block diagram of the differential encoder is shown in Figure 2.1.3.3.1-2. The binary data stream entering the modulator, b_m, is converted by a serial-to-parallel converter into two separate binary streams (X_k) and (Y_k). Starting from bit 1 in time of stream b_m, all odd numbered bits form stream X_k and all even numbered bits form stream Y_k.

Figure 2.1.3.3.1-2 Differential Encoder



The digital data sequences (X_k) and (Y_k) are encoded onto (I_k) and (Q_k) according to:

$$I_k = I_{k-1} \cos[\Delta\Phi(X_k,Y_k)] - Q_{k-1} \sin[\Delta\Phi(X_k,Y_k)]$$

$$Q_k = I_{k-1} \sin[\Delta\Phi(X_k,Y_k)] + Q_{k-1} \cos[\Delta\Phi(X_k,Y_k)]$$

where I_{k-1}, Q_{k-1} are the amplitudes at the previous pulse time. The phase change $\Delta\Phi$ is determined according to the following table:

X_k	Y_k	$\Delta\Phi$
1	1	$\frac{-3\pi}{4}$
0	1	$\frac{3\pi}{4}$
0	0	$\frac{\pi}{4}$
1	0	$\frac{-\pi}{4}$

The signals I_k,Q_k at the output of the differential phase encoding block can take one of five values, 0, ± 1, $\pm \frac{1}{\sqrt{2}}$, resulting in the constellation shown in Figure 2.1.3.3.1-1.

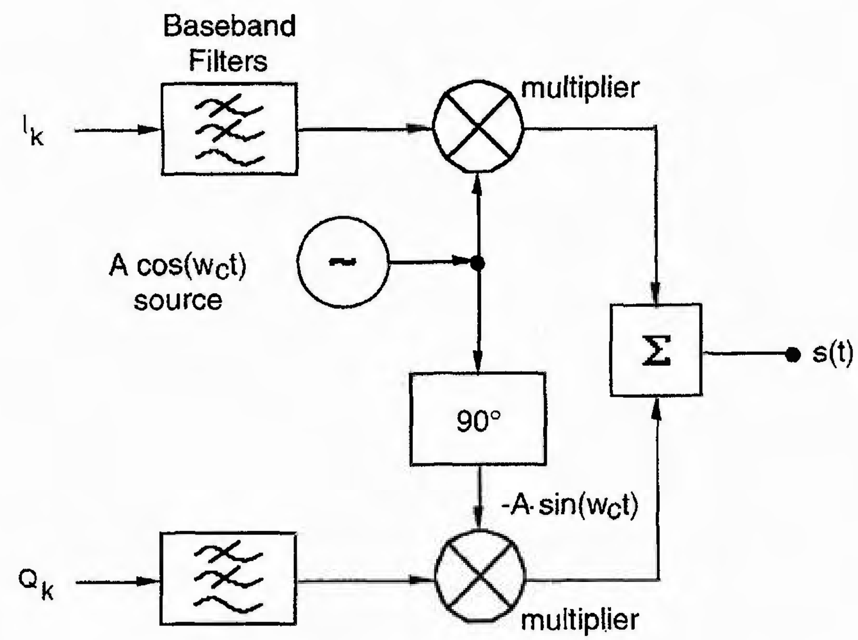
Impulses I_k, Q_k are applied to the inputs of the I & Q base-band filters. The base-band filters shall have linear phase and square root raised cosine frequency response of the form:

$$|H(f)| = \begin{cases} 1 & 0 \le f \le \frac{(1-\alpha)}{2T} \\ \sqrt{\frac{1}{2}\left\{1 - \sin\left[\frac{\pi(2fT-1)}{2\alpha}\right]\right\}} & \frac{(1-\alpha)}{2T} \le f \le \frac{(1+\alpha)}{2T} \\ 0 & f > \frac{(1+\alpha)}{2T} \end{cases}$$

where T is the symbol period. The roll-off factor, α, determines the width of the transition band, and is 0.35.

Figure 2.1.3.3.1-3 is for explanatory purposes and does not prescribe a specific implementation.

Figure 2.1.3.3.1-3 Transmit Signal Generation



The resultant transmitted signal s(t) is given by :

$$s(t) = \sum_n g(t - nT)\cos\Phi_n \cos\omega_c t - \sum_n g(t - nT)\sin\Phi_n \sin\omega_c t$$

where g(t) is the pulse shaping function, ω_c is the radian carrier frequency, T is the symbol period, and Φ_n is the absolute phase corresponding to the n^{th} symbol interval.

The Φ_n which results from the differential encoding is:

$$\Phi_n = \Phi_{n-1} + \Delta\Phi_n.$$

Any method which generates the specified s(t) using the cited phase table may be used.

2.1.3.3.1.1 Modulation Accuracy

2.1.3.3.1.1.1 Description Of The Technique Used To Specify The Modulation Accuracy Requirement

The modulation accuracy requirement is specified by setting limits on the RMS difference between the actual transmitted signal waveform and the ideal signal waveform. The ideal waveform is derived mathematically from the specification of modulation in Section 2.1.3.3.1. The specified requirement is error vector magnitude.

2.1.3.3.1.1.2 Average Frequency Error Definition

For this measurement, frequency accuracy shall meet the requirements of Section 2.1.1.2.2. (and Section 3.1.1.2.2) prior to measurement.

The average carrier frequency error is the difference between the average carrier frequency of the actual transmitted waveform and the average signal waveform carrier frequency.

2.1.3.3.1.1.3 Error Vector Magnitude Requirement

The ideal modulation is defined in Section 2.1.3.3.1. The definition is such that, observing an ideal transmitter through an ideal root raised-cosine receiver filter at the correct sampling instants one symbol apart would result in the sequence of values given by:

$$S(k) = S(k-1)e^{j[\pi/4 + B(k) \bullet \pi/2]}$$

where B(k) = 0,1,2, 3 according to the following table:

X_k	Y_k	B(k)
0	0	0
0	1	1
1	1	2
1	0	3

In the forward channel, S(k) forms part of a continuous data stream. In the reverse channel, the transmit bursts from the mobile station are truncated by power up and down ramping. In this case, S(6) is the first sample that enters into demodulation, which yields the first two information bits by comparing S(6) with S(7). The last two information bits lie in the comparison of S(162) and S(161).

The ideal transmit and receive filters in cascade form a raised cosine Nyquist filter having an impulse response going through zero at symbol period intervals, so there is no inter-symbol interference at the ideal sampling points. The ideal signal samples therefore, take on one of the eight values defined above, at the output of the receive filter.

This section defines how the output signal from a transmitter is to be evaluated against the ideal signal.

Let Z(k) be the complex vectors produced by observing the real transmitter through an ideal measuring receiver filter at instants k, one symbol period apart. With S(k) defined as above, the transmitter is modeled as:

$$Z(k) = [C0 + C1 \bullet [S(k) + E(k)]] \bullet W^k$$

where:

$W = e^{dr + jda}$ accounts for both a frequency offset giving "da" radians per symbol phase rotation and an amplitude changes of "dr" nepers per symbol;

C0 is a constant origin offset representing quadrature modulator imbalance,

C1 is a complex constant representing the arbitrary phase and output power of the transmitter, and

E(k) is the residual vector error on sample S(k).

The sum square vector error is then:

$$\sum_{k=MIN}^{k=MAX} |E(k)|^2 = \sum_{k=MIN}^{k=MAX} \left| \left\{ \left[Z(k) \bullet W^{-k} - C0 \right] / C1 \right\} - S(k) \right|^2 ; \; |S(k)| = 1$$

C0, C1 and W shall be chosen to minimize this expression and are then used to compute the individual vector errors E(k) on each symbol. The symbol timing phase of the receiver output samples used to compute the vector error shall also be chosen to give the lowest value.

The values of MAX and MIN for the reverse channel (mobile station transmitter) are:

MIN = 6 (the vector in the last of the three ramp-up symbol periods)

MAX = 162 (the vector in the first of the three ramp-down symbol periods)

The RMS vector error is then computed as the square root of the sum-square vector divided by the number of symbols in the slot, 162 in the forward direction and 157 in the reverse direction.

The RMS vector error in any burst shall be less than 0.125.

In addition, the normalized error vector magnitude during the first 10 symbols (20 bits) of a mobile station TDMA burst following the ramp-up, must have an RMS value of less than 0.25 when averaged over 10 bursts within a 1 minute interval.

Note: The value of origin offset for minimum performance referred to in Section 2.1.3.3.1.1.3 will be addressed, if deemed necessary, in IS-137 and IS-138.

2.1.3.3.2 Speech Coding (Full-Rate)

See Section 3 in IS-136.1.

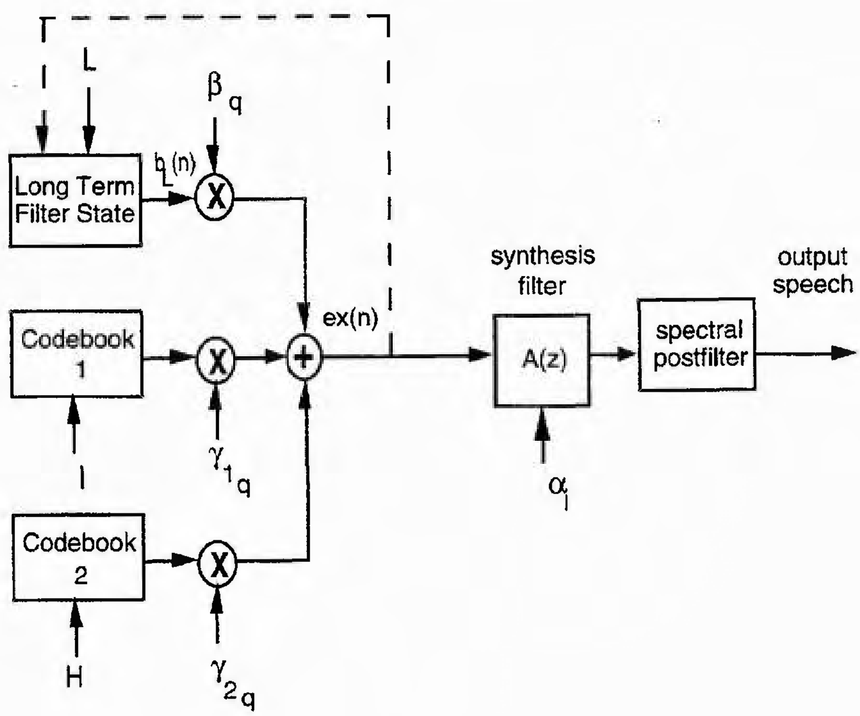
The speech coding algorithms for alternate codecs are defined in IS-641.

The speech coding algorithm described in this standard (see Sections 2.1.3.3.2.1 to 2.1.3.3.2.6) is a member of a class of speech codecs known as Code Excited Linear Predictive Coding (CELP), Stochastic Coding or Vector Excited Speech Coding. These techniques use codebooks to vector quantize the excitation (residual) signal. The speech coding algorithm is a variation on CELP called Vector-Sum Excited Linear Predictive Coding (VSELP). VSELP uses a codebook which has a predefined structure such that the computations required for the Codebook Search process can be significantly reduced.

2.1.3.3.2.1 Definitions and Basic Codec Parameters

Figure 2.1.3.3.2.1–1 shows a block diagram of the speech decoder. This figure indicates the various parameters which must be determined and encoded by the speech codec.

Figure 2.1.3.3.2.1–1 Speech Decoder



The speech decoder utilizes two VSELP excitation codebooks. The two codebooks each have their own gain. The two codebook excitations are each multiplied by their corresponding gains and summed to create a combined codebook excitation. The following are the basic parameters for the 7950 bps speech codec and decoder.

	Sampling Rate	**8 kHz**
N_F	frame length	160 samples (20 msec)
N	subframe length	40 samples (5 msec)
N_P	short term predictor order	10
	# of taps for long term predictor	1
M_1	# of bits in codeword 1 (# of basis vectors)	7
M_2	# of bits in codeword 2 (# of basis vectors)	7

The basic data rate of the speech codec is 7950 bps. There are 159 bits per speech frame (20 msec) for the speech codec. These 159 bits are allocated as follows:

short-term filter coefficients, α_i's		38 bits/frame
frame energy, R(0)		5 bits/frame
lag, L	7 bits/subframe	28 bits/frame
codewords, I, H	7+7 bits/subframe	56 bits/frame
gains β, γ_1, γ_2	8 bits/subframe	32 bits/frame

The following is a list of all the parameter codes transmitted for each 20 msec. speech frame. The codes are:

R0	5 bits	frame energy
LPC1	6 bits	1st reflection coefficient
LPC2	5 bits	2nd reflection coefficient
LPC3	5 bits	3rd reflection coefficient
LPC4	4 bits	4th reflection coefficient
LPC5	4 bits	5th reflection coefficient
LPC6	3 bits	6th reflection coefficient
LPC7	3 bits	7th reflection coefficient
LPC8	3 bits	8th reflection coefficient
LPC9	3 bits	9th reflection coefficient
LPC10	2 bits	10th reflection coefficient
LAG_1	7 bits	lag for first subframe
LAG_2	7 bits	lag for second subframe
LAG_3	7 bits	lag for third subframe
LAG_4	7 bits	lag for fourth subframe
CODE1_1	7 bits	1st codebook code, I, for first subframe
CODE1_2	7 bits	1st codebook code, I, for second subframe

CODE1_3	7 bits	1st codebook code, I, for third subframe
CODE1_4	7 bits	1st codebook code, I, for fourth subframe
CODE2_1	7 bits	2nd codebook code, H, for first subframe
CODE2_2	7 bits	2nd codebook code, H, for second subframe
CODE2_3	7 bits	2nd codebook code, H, for third subframe
CODE2_4	7 bits	2nd codebook code, H, for fourth subframe
GSP0_1	8 bits	{GS, P0, P1} code for first subframe
GSP0_2	8 bits	{GS, P0, P1} code for second subframe
GSP0_3	8 bits	{GS, P0, P1} code for third subframe
GSP0_4	8 bits	{GS, P0, P1} code for fourth subframe

2.1.3.3.2.2 Audio Interface

Due to the delays inherent in the air interface specification, which may exceed 100 msec, the implementer is cautioned that echo control measures are necessary.

The function of the audio interface at the mobile station transmitter is to convert the analog speech signal to uniform PCM format with a minimum resolution of 13 bits for further processing by the speech codec.

The speech codec is preceded by the following voice processing stages:

- Level adjustment
- Bandpass Filter
- Analog to Digital Converter.

The characteristics of these stages are described in the following sections.

2.1.3.3.2.2.1 Transmit Level Adjustment

Pending the generation of a complete speech transmission plan for these systems, the following requirements shall be met to ensure compatibility with the transmission plan for fixed digital speech networks.

The transmit audio sensitivity of the mobile station shall meet the requirements stated in IS-137 under "Digital Transmitter Audio Sensitivity".

2.1.3.3.2.2.2 Bandpass Filter

The function of the filter is to avoid aliasing distortion of the input signal. The attenuation of the filter shall comply with ITU (formerly CCITT) Red Book G.714 sending filter.

2.1.3.3.2.2.3 Analog to Digital Converter

The A/D function shall be performed according to either of the following:

- by direct conversion analog to a uniform PCM format with a minimum resolution of 13 bits,
- or by converting analog to an 8-bit/μlaw format followed by a μlaw/uniform code conversion.

The A/D conversion is based on the standard 8-bit/μlaw codec specified in ITU (formerly CCITT) Red Book G.711.

The μlaw/uniform code conversion is performed according to definition in ITU (formerly CCITT) Red Book G.721 Section 4.2.1 sub-block EXPAND. The parameter LAW shall be set to LAW = 0.

2.1.3.3.2.2.4 Echo Return Loss

The echo return loss of the mobile station shall have a minimum value of 45 dB, measured in accordance with the procedure given in IEEE 269-1990, Section 7.10.2, condition 3, using the continuous-spectrum signal method; and calculated using the technique described in TIA SP-1920A, Section 4.4.1.1. This requirement must be met by all types of mobile stations at their nominal volume setting.

2.1.3.3.2.3 Pre-Processing

It may be desirable in some instances to provide additional high-pass filtering after analog to digital conversion. A fourth order Chebyshev type II high-pass with a filter response which is 3 dB down at 120 Hz and 40 dB down at 60 Hz may be used. The transfer function of the high-pass filter is given by:

$$H_{hp}(z) = \frac{\sum_{i=0}^{4} a_i z^{-i}}{1 - \sum_{i=1}^{4} b_i z^{-i}} \qquad (2.1.3.3.2.3\text{–}1)$$

where:

a_0 =	0.898025036		
a_1 =	–3.59010601	b_1 =	3.78284979
a_2 =	5.38416243	b_2 =	–5.37379122
a_3 =	–3.59010601	b_3 =	3.39733505
a_4 =	0.898024917	b_4 =	–0.806448996

2.1.3.3.2.4 Short-Term Predictor Coefficients

The short-term filter is equivalent to the traditional LPC synthesis filter. The transfer function for the short-term filter is given by:

$$A(z) = \frac{1}{1-\sum_{i=1}^{N_p} \alpha_i z^{-i}} \quad (2.1.3.3.2.4\text{–}1)$$

The short term predictor parameters are the α_i's of the short term or synthesis filter. These are standard LPC direct form filter coefficients. The short term predictor parameters are computed from the input speech. No pre-emphasis is used. This analysis interval should be centered with respect to the center of the fourth subframe of each frame. The order of the predictor is 10 (Np = 10).

2.1.3.3.2.4.1 Solution for Reflection Coefficients

An efficient fixed point covariance lattice algorithm, FLAT, may be used for determination of the short-term filter coefficients. Let the samples of the input speech which fall in the analysis interval be represented by s(n); $0 \le n \le N_A-1$. The analysis length used for computation of the parameters is 170 samples ($N_A = 170$).

Since FLAT is a lattice algorithm one can view the technique as trying to build an optimum (that which minimizes residual energy) inverse lattice filter stage by stage.

Defining $b_j(n)$ to be the backward residual out of stage j of the inverse lattice filter and $f_j(n)$ to be the forward residual out of stage j of the inverse lattice filter we can define:

$$F_j(i,k) = \sum_{n=N_P}^{N_A-1} f_j(n-i)f_j(n-k) \quad (2.1.3.3.2.4.1\text{–}1)$$

the autocorrelation of $f_j(n)$;

$$B_j(i,k) = \sum_{n=N_P}^{N_A-1} b_j(n-i-1)b_j(n-k-1) \quad (2.1.3.3.2.4.1\text{–}2)$$

the autocorrelation of $b_j(n-1)$ and:

$$C_j(i,k) = \sum_{n=N_P}^{N_A-1} f_j(n-i)b_j(n-k-1) \quad (2.1.3.3.2.4.1\text{–}3)$$

the cross correlation between $f_j(n)$ and $b_j(n-1)$.

Let r_j represent the reflection coefficient for stage j of the inverse lattice. Then:

$$F_j(i,k) = F_{j-1}(i,k) + r_j\left(C_{j-1}(i,k) + C_{j-1}(k,i)\right) + r_j^2 B_{j-1}(i,k) \qquad (2.1.3.3.2.4.1\text{–}4)$$

and

$$B_j(i,k) = B_{j-1}(i+1,k+1) + r_j\left(C_{j-1}(i+1,k+1) + C_{j-1}(k+1,i+1)\right) + r_j^2 F_{j-1}(i+1,k+1)$$

(2.1.3.3.2.4.1-5)

and

$$C_j(i,k) = C_{j-1}(i,k+1) + r_j\left(B_{j-1}(i,k+1) + F_{j-1}(i,k+1)\right) + r_j^2 C_{j-1}(k+1,i)$$

(2.1.3.3.2.4.1-6)

r_j can be expressed as:

$$r_j = -2\frac{C_{j-1}(0,0) + C_{j-1}(N_P - j, N_P - j)}{F_{j-1}(0,0) + B_{j-1}(0,0) + F_{j-1}(N_P - j, N_P - j) + B_{j-1}(N_P - j, N_P - j)}$$

(2.1.3.3.2.4.1–7)

The FLAT algorithm can now be stated as follows:

1. First compute the covariance (autocorrelation) matrix from the input speech:

$$\phi(i,k) = \sum_{n=N_P}^{N_A-1} s(n-i)\, s(n-k) \qquad (2.1.3.3.2.4.1\text{–}8)$$

   for $0 \le i,k \le Np$.

   (Note: see also Section 2.1.3.3.2.4.2).

2. | | | |
   |---|---|---|
   | $F_0(i,k) = \phi(i,k)$ | $0 \le i,k \le Np-1$ | (2.1.3.3.2.4.1–9) |
   | $B_0(i,k) = \phi(i+1,k+1)$ | $0 \le i,k \le Np-1$ | (2.1.3.3.2.4.1–10) |
   | $C_0(i,k) = \phi(i,k+1)$ | $0 \le i,k \le Np-1$ | (2.1.3.3.2.4.1–11) |

3. set j = 1

4. Compute r_j using (2.1.3.3.2.4.1–7)

5. Quantize r_j (see Section 2.1.3.3.2.4.3)

6. If j = Np then done.

7. | | | |
   |---|---|---|
   | Compute $F_j(i,k)$ | $0 \le i,k \le Np-j-1$ | using (2.1.3.3.2.4.1–4) |
   | Compute $B_j(i,k)$ | $0 \le i,k \le Np-j-1$ | using (2.1.3.3.2.4.1–5) |
   | Compute $C_j(i,k)$ | $0 \le i,k \le Np-j-1$ | using (2.1.3.3.2.4.1–6) |

8. j = j+1; go to 4.

This algorithm can be simplified by noting that the ϕ, F and B matrices are symmetric such that only the upper triangular part of the matrices need to be computed or updated. In addition $\phi(i,j)$ can be efficiently computed from $\phi(i-1,j-1)$. Also, if step 7 is done so that $F_j(i,k)$, $B_j(i-1,k-1)$, $C_j(i,k-1)$, and $C_j(k,i-1)$ are updated together, then common terms can be computed once and the recursion can be done in place.

2.1.3.3.2.4.2 Bandwidth Expansion

The speech codec should provide for a small amount of bandwidth expansion of the short-term filter coefficients prior to quantization. Windowing of the autocorrelations prior to the solution of the reflection coefficients is one technique which may be used.

Prior to solving for the reflection coefficients, the ϕ array is modified by windowing the autocorrelation functions:

$$\phi'(i,k) = \phi(i,k)w(|i-k|) \qquad (2.1.3.3.2.4.2\text{–}1)$$

The window used is a binomial window with an effective bandwidth of 80 Hz. The values of w(i) are:

w(0)	1.000000
w(1)	0.999644
w(2)	0.998577
w(3)	0.996802
w(4)	0.994321
w(5)	0.991141
w(6)	0.987268
w(7)	0.982710
w(8)	0.977478
w(9)	0.971581
w(10)	0.965032

(2.1.3.3.2.4.2–2)

2.1.3.3.2.4.3 Quantization and Encoding of Coefficients

As stated in Section 2.1.3.3.2.4.1, the quantization of the reflection coefficients may be done within the FLAT recursion.

The reflection coefficients are quantized using codebooks designed for each reflection coefficient. The bit allocations for the reflection coefficient quantizers are:

r_1	6 bits
r_2	5 bits
r_3	5 bits
r_4	4 bits
r_5	4 bits
r_6	3 bits
r_7	3 bits
r_8	3 bits
r_9	3 bits
r_{10}	2 bits

The 10 reflection coefficients ($r_1 - r_{10}$) which represent the short term predictor parameters are each independently quantized. Table 2.1.3.3.2.4.3–1 provides the 10 codebooks for quantization of the 10 reflection coefficients to provide codes LPC1 through LPC10. The quantization may be performed by finding the codebook value which is closest (minimum absolute error) from the unquantized reflection coefficient. The code for that codebook entry (which appears in the left column) is the code for that reflection coefficient. Note that no transformation (such as log area ratio or arc sine) of the reflection coefficients is required during the quantization process.

Table 2.1.3.3.2.4.3–1 Codebooks for Quantization of Coefficients

64 QUANTIZATION LEVELS FOR r_1 FOLLOW:

0	–0.9867044091E+00	32	–0.6262212992E+00
1	–0.9810330868E+00	33	–0.5971570611E+00
2	–0.9762308002E+00	34	–0.5684631467E+00
3	–0.9711073637E+00	35	–0.5378258228E+00
4	–0.9655630589E+00	36	–0.5058867931E+00
5	–0.9597059488E+00	37	–0.4740323126E+00
6	–0.9536622763E+00	38	–0.4414438009E+00
7	–0.9471911192E+00	39	–0.4054017663E+00
8	–0.9406521916E+00	40	–0.3682330251E+00
9	–0.9339897037E+00	41	–0.3293099701E+00
10	–0.9266145825E+00	42	–0.2894666791E+00
11	–0.9190770388E+00	43	–0.2428349406E+00
12	–0.9110740423E+00	44	–0.1948891282E+00
13	–0.9032388926E+00	45	–0.1466629505E+00
14	–0.8951876163E+00	46	–0.9152601659E–01
15	–0.8865973353E+00	47	–0.2692181431E–01
16	–0.8775991797E+00	48	0.3727672249E–01
17	–0.8679655194E+00	49	0.1093522757E+00
18	–0.8578169942E+00	50	0.1758577228E+00
19	–0.8468435407E+00	51	0.2397777289E+00
20	–0.8350597620E+00	52	0.3001485765E+00
21	–0.8232460022E+00	53	0.3555985391E+00
22	–0.8109311461E+00	54	0.4108347893E+00
23	–0.7979614735E+00	55	0.4679426551E+00
24	–0.7842678428E+00	56	0.5202128887E+00
25	–0.7699828744E+00	57	0.5746787190E+00
26	–0.7545526624E+00	58	0.6337370872E+00
27	–0.7377604842E+00	59	0.6966888309E+00
28	–0.7188396454E+00	60	0.7613552213E+00
29	–0.6990101337E+00	61	0.8211135268E+00
30	–0.6768131256E+00	62	0.8759805560E+00
31	–0.6533866525E+00	63	0.9311733246E+00

32 QUANTIZATION LEVELS FOR r_2 FOLLOW:

0	–0.7547348738E+00	16	0.5302551389E+00
1	–0.5826729536E+00	17	0.5760165453E+00
2	–0.4569368660E+00	18	0.6193220615E+00
3	–0.3481135964E+00	19	0.6593915224E+00
4	–0.2492762953E+00	20	0.6967787147E+00
5	–0.1585778296E+00	21	0.7315257788E+00
6	–0.7726432383E–01	22	0.7650170326E+00
7	–0.5096863955E–02	23	0.7966732979E+00
8	0.6527176499E–01	24	0.8267812133E+00
9	0.1329884380E+00	25	0.8543012142E+00
10	0.1978287548E+00	26	0.8798117638E+00
11	0.2600678802E+00	27	0.9037305117E+00
12	0.3186267912E+00	28	0.9251338840E+00
13	0.3747462034E+00	29	0.9448361397E+00
14	0.4288900495E+00	30	0.9636774063E+00
15	0.4810178876E+00	31	0.9816107750E+00

32 QUANTIZATION LEVELS FOR r_3 FOLLOW:

0	–0.8606231213E+00	16	–0.1760748774E+00
1	–0.8046579361E+00	17	–0.1370347440E+00
2	–0.7523136735E+00	18	–0.9637858719E–01
3	–0.7056827545E+00	19	–0.5570860580E–01
4	–0.6582847834E+00	20	–0.1342663728E–01
5	–0.6130494475E+00	21	0.2913235873E–01
6	–0.5684005022E+00	22	0.7243801653E–01
7	–0.5247684717E+00	23	0.1183170006E+00
8	–0.4832728207E+00	24	0.1668847799E+00
9	–0.4436871707E+00	25	0.2185972333E+00
10	–0.4048590660E+00	26	0.2741918266E+00
11	–0.3659544587E+00	27	0.3353714645E+00
12	–0.3276270330E+00	28	0.4032742083E+00
13	–0.2901176810E+00	29	0.4797808230E+00
14	–0.2521926463E+00	30	0.5761802793E+00
15	–0.2139451057E+00	31	0.6969622374E+00

1 16 QUANTIZATION LEVELS FOR r_4 FOLLOW:

0	–0.4505536556E+00	8	0.3864109516E+00
1	–0.2582354248E+00	9	0.4573097229E+00
2	–0.1276191175E+00	10	0.5282919407E+00
3	–0.1891620830E–01	11	0.5986876488E+00
4	0.7355944812E–01	12	0.6679506302E+00
5	0.1581759751E+00	13	0.7368153930E+00
6	0.2381238639E+00	14	0.8054329753E+00
7	0.3138437271E+00	15	0.8715001345E+00

2 16 QUANTIZATION LEVELS FOR r_5 FOLLOW:

0	–0.6570090652E+00	8	–0.1441930979E–01
1	–0.5339469314E+00	9	0.4928108677E–01
2	–0.4366474450E+00	10	0.1149446666E+00
3	–0.3503953516E+00	11	0.1853068322E+00
4	–0.2729992270E+00	12	0.2629969418E+00
5	–0.2044571489E+00	13	0.3558021188E+00
6	–0.1395732164E+00	14	0.4658669233E+00
7	–0.7738068700E–01	15	0.6091661453E+00

3 8 QUANTIZATION LEVELS FOR r_6 FOLLOW:

0	–0.3351013660E+00	4	0.2685563564E+00
1	–0.1157702208E+00	5	0.3836385608E+00
2	0.3265872598E–01	6	0.5067840815E+00
3	0.1545225680E+00	7	0.6525010467E+00

4 8 QUANTIZATION LEVELS FOR r_7 FOLLOW:

0	–0.5834863186E+00	4	–0.5848890170E–01
1	–0.4363777936E+00	5	0.5718512833E–01
2	–0.3007811308E+00	6	0.1880214661E+00
3	–0.1750739664E+00	7	0.3601035774E+00

8 QUANTIZATION LEVELS FOR r_8 FOLLOW:

0	–0.3832900822E+00	4	0.1637183726E+00
1	–0.2111644298E+00	5	0.2812094688E+00
2	–0.7242984325E–01	6	0.4174390733E+00
3	0.4901395738E–01	7	0.5824319720E+00

8 QUANTIZATION LEVELS FOR r_9 FOLLOW:

0	–0.5373110771E+00	4	–0.2935140580E–01
1	–0.3927696943E+00	5	0.8255130798E–01
2	–0.2594878078E+00	6	0.2107033432E+00
3	–0.1391013712E+00	7	0.3958446980E+00

4 QUANTIZATION LEVELS FOR r_{10} FOLLOW:

0	–0.1508204788E+00	2	0.1958054304E+00
1	0.4652437568E–01	3	0.3621688187E+00

2.1.3.3.2.4.4 (Intentionally Left Blank)

2.1.3.3.2.4.5 Conversion to Direct Form Coefficients

The quantized reflection coefficients must be converted to direct form filter coefficients, α_i's.

2.1.3.3.2.4.6 Interpolation of Coefficients

The speech codec linearly interpolates the α_i's for the first, second and third subframes of each frame. The fourth subframe uses the uninterpolated α_i's for that frame. For all α_i's:

$$\alpha_i = .75\alpha_{i(previous)} + .25\alpha_{i(current)} \quad \text{for subframe 1}$$
$$\alpha_i = .5\alpha_{i(previous)} + .5\alpha_{i(current)} \quad \text{for subframe 2}$$
$$\alpha_i = .25\alpha_{i(previous)} + .75\alpha_{i(current)} \quad \text{for subframe 3}$$
$$\alpha_i = \alpha_{i(current)} \quad \text{for subframe 4}$$

(2.1.3.3.2.4.6–1)

where $\alpha_{i(previous)}$ is the i^{th} direct form coefficient from the previous frame and $\alpha_{i(current)}$ is the i^{th} direct form coefficient from the current frame.

For interpolated subframes, the α_i's are converted to reflection coefficients to check for filter stability. If the resulting filter is unstable (any reflection coefficient having a magnitude equal to or greater than 1.0) then uninterpolated coefficients are used for that subframe. The uninterpolated coefficients used for subframe 1 are the previous frame's coefficients. The uninterpolated coefficients used for subframe 3 are the current frame's coefficients. Subframe 2 uses either the previous frame or the current frame's coefficients, choosing those coefficients which correspond to the frame with the higher energy. If both frames have the same energy (R(0)) then subframe 2 uses the previous frame's coefficients.

2.1.3.3.2.5 Frame Energy

An energy value is computed and encoded once per frame. This energy value, R(0), reflects the average signal power in the input speech over a 20 msec. interval which is centered with respect to the middle of the fourth subframe.

2.1.3.3.2.5.1 Computation of Frame Energy

R(0), may be computed during the computation of the short term predictor parameters.

$$R(0) = \frac{\phi(0,0)+\phi(N_P,N_P)}{2\,(N_A - N_P)} \qquad (2.1.3.3.2.5.1\text{–}1)$$

where $\phi(i,k)$ is defined by (2.1.3.3.2.4.1–8). Equation 2.1.3.3.2.5.1–1 can be rewritten as:

$$R(0) = \frac{\phi(0,0)+\phi(10,10)}{320} \qquad (2.1.3.3.2.5.1\text{–}2)$$

2.1.3.3.2.5.2 Quantization and Encoding of Frame Energy

R(0) is converted into dB relative to full scale (full scale, R_{max}, is defined as the square of the maximum sample amplitude).

$$R_{dB} = 10 \log_{10}(R(0) / R_{max}) \qquad (2.1.3.3.2.5.2\text{–}1)$$

R_{dB} is then quantized to 32 levels. A code of zero for R0 corresponds to an energy of 0 (R(0) = 0). This code can be used to totally silence the speech decoder. The remaining 31 quantized values for R_{dB} range from a minimum of –64 (corresponding to a code of 1 for R0) to a maximum of –4 (corresponding to a code of 31 for R0). The step size of the quantizer is 2 (2 dB steps). R0 is chosen as:

$$\text{minimize (for R0=1 to 31) } [\text{abs}(R0-(R_{dB} + 66)/2)] \quad \text{if } R_{dB} \geq -72 \qquad (2.1.3.3.2.5.2\text{–}2a)$$

$$\text{else } R0 = 0 \quad \text{if } R_{dB} < -72 \qquad (2.1.3.3.2.5.2\text{–}2b)$$

where R0 can take on the integer values from 0 to 31 corresponding to the 32 codes for R0.

The quantized value of R(0), $R_q(0)$ is given by:

$$R_q(0) = R_{max}(10.\ {**}\ (((2 * R0) - 66) / 10)) \quad \text{if } R0 \neq 0 \qquad (2.1.3.3.2.5.2\text{–}3a)$$

$$R_q(0) = 0 \quad \text{if } R0 = 0 \qquad (2.1.3.3.2.5.2\text{–}3b)$$

2.1.3.3.2.5.3 Interpolation of Frame Energy

Define $R'_q(0)$ to be the quantized value of R(0) to be used for the subframe and $R_q(0)$ to be the quantized value of R(0). Then:

$$R'_q(0) = R_q(0)_{\text{previous frame}} \quad \text{for subframe 1} \qquad (2.1.3.3.2.5.3\text{–}1a)$$

$$R'_q(0) = R_q(0)_{\text{current frame}} \quad \text{for subframes 3 and 4} \qquad (2.1.3.3.2.5.3\text{–}1b)$$

$$R'_q(0) = \sqrt{R_q(0)_{\text{previous frame}} \; R_q(0)_{\text{current frame}}} \quad \text{for subframe 2} \qquad (2.1.3.3.2.5.3\text{–}1c)$$

2.1.3.3.2.6 Subframe Processing

The 20 msec. speech frame is subdivided into four 5 msec. subframes. For each subframe the speech codec must determine and code the long-term predictor lag, L, the two codewords, I and H, and the gains, β, γ_1 and γ_2.

2.1.3.3.2.6.1 Weighting of Input Speech

The speech codec utilizes a perceptual noise weighting filter of the form:

$$W(z) = \frac{1-\sum_{i=1}^{N_p} \alpha_i z^{-i}}{1-\sum_{i=1}^{N_p} \alpha_i \lambda^{i} z^{-i}} \qquad (2.1.3.3.2.6.1\text{–}1)$$

where α_i's are the filter coefficients for the subframe (see Section 2.1.3.3.2.4.6) and λ is the noise weighting parameter. A value of $\lambda = .8$ may be used. The input speech for the subframe must be filtered by the perceptual weighting filter. The filter may be implemented as a cascade of a 10^{th} order inverse (all zero) direct form filter corresponding to the numerator of equation 2.1.3.3.2.6.1–1, followed by a 10^{th} order (all pole) direct form filter corresponding to the denominator. The filter coefficients will change for each subframe. The states of the filters should be preserved from one subframe to the next. (Note: Other weighting filter implementations may adversely affect performance. Consequently, care must be taken to achieve or exceed the performance level that is realized by the above filter configuration.)

2.1.3.3.2.6.2 Subtraction of Zero Input Response

The speech encoder is an analysis by synthesis coding system. Therefore a version of the speech decoder is used in the speech encoder. The form of the synthesis filter used in the speech encoder is given by:

$$H(z) = \frac{1}{1 - \sum_{i=1}^{N_p} \alpha_i \lambda^i z^{-i}} \qquad (2.1.3.3.2.6.2\text{–}1)$$

Note that the synthesis filter used in the speech encoder is different than that used in the speech decoder. The synthesis for the speech encoder includes the same noise weighting parameter, λ, as the weighting filter in Section 2.1.3.3.2.6.1. The synthesis filter used in the speech codec is therefore called the weighted synthesis filter. A weighted synthesis filter is used to match the weighting applied to the input speech. The weighted synthesis filter will have a filter state associated with it at the start of each subframe. In order to remove the effects of the weighted synthesis filter's initial state from the subframe parameter determinations, the zero input response of the weighted synthesis filter shall be computed and subtracted from the weighted input speech for the subframe. The weighted synthesis filter should be implemented with a direct form filter.

2.1.3.3.2.6.3 Long Term Predictor Lag

The speech codec uses the closed loop approach to choosing the long term predictor lag. In the closed loop case the lag is determined from only past output from the long term filter and the current input speech. The long term filter response can be expressed as:

$$B_n(z) = \frac{1}{1 - \beta z^{-\left\lfloor \frac{n+L}{L} \right\rfloor L}} \qquad (2.1.3.3.2.6.3\text{–}1)$$

where $\lfloor x \rfloor$ is the floor function of x which evaluates to the largest integer $\leq x$ and n is the sample in the subframe; $0 \leq n \leq N-1$. The lag L is always used as the delay for the first L samples of the subframe. If $L < N$ then for samples $n = L$ to $2L-1$, a delay of $2L$ is used, etc. In this way the delay is always greater than n so that only long term filter states existing at the start of the subframe are used.

Figure 2.1.3.3.2.6.3–1 Analysis by Synthesis Procedure for Long Term Predictor Lag and Code Search

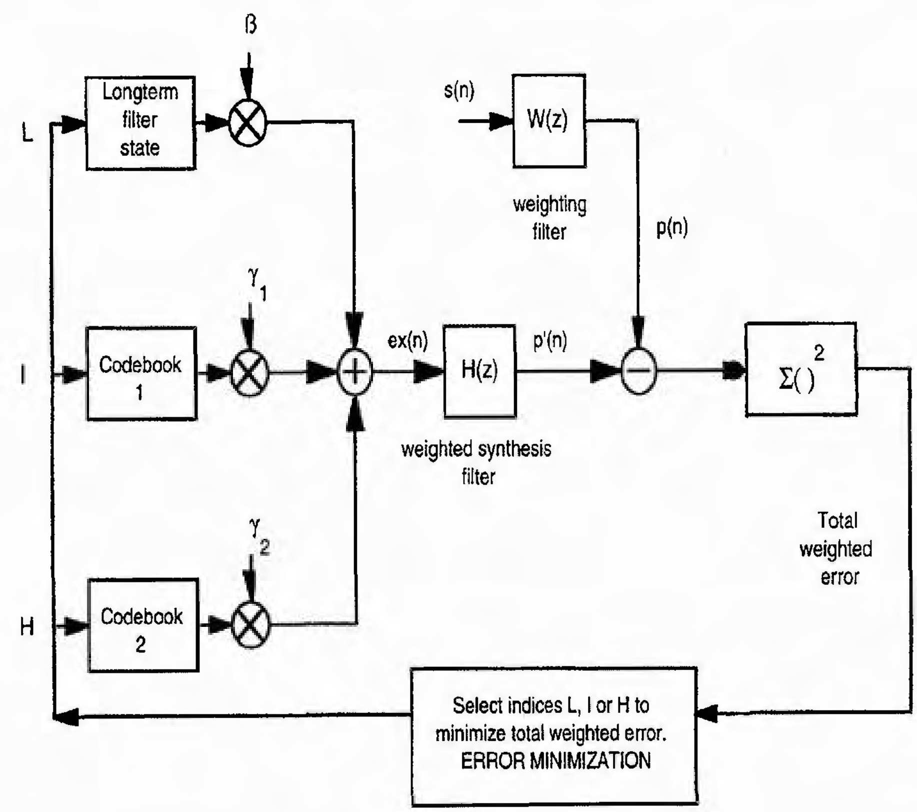


Figure 2.1.3.3.2.6.3–1 illustrates how the long term predictor lag search and code search can be formulated utilizing the long term filter just described. The input p(n) in this case is the weighted input speech for the subframe minus the zero input response of the weighted synthesis filter, H(z). The long term lag optimization looks just like a codebook search where the codebook is defined by the long term filter state and the specific vector in the codebook is specified by the long term predictor lag, L. These three "codebooks" (the long term predictor state and the two excitation codebooks) are searched sequentially. The long term predictor lag is determined first assuming no input from the excitation codebooks.

2.1.3.3.2.6.3.1 Computation of Lag

Defining:

- L_{min} minimum possible value for long term lag L ($L_{min} = 20$)
- $r(n)$ long term filter state; $n < 0$ (past outputs of long term filter)
- $b_L(n)$ output of long term filter state "codebook" for lag L
- $h(n)$ impulse response of $H(z)$
- $b'_L(n)$ $b_L(n)$ filtered by $H(z)$ (convolved with $h(n)$)

$$G_L = \sum_{n=0}^{N-1} \left(b'_L(n)\right)^2 \qquad (2.1.3.3.2.6.3.1\text{–}1)$$

$$C_L = \sum_{n=0}^{N-1} b'_L(n)\, p(n) \qquad (2.1.3.3.2.6.3.1\text{–}2)$$

Then the lag, L, which will minimize the total weighted error (with optimal β) should be chosen as that which maximizes

$$(C_L)^2 / G_L \qquad (2.1.3.3.2.6.3.1\text{–}3)$$

Since we are restricting β to be positive, only lags with positive C_L are considered. L is coded with 7 bits and can take on a value from 20 to 146. One of the 128 coded lag values is reserved to indicate that the long term predictor is not used. This allows the long term predictor to be disabled when a positive correlation cannot be found. In order for this search to be efficient, $b'_L(n)$ must be computed efficiently. This may be done as follows:

define:

$$z_L(n) = \sum_{i=0}^{\min(n,L-1)} r(i-L)\, h(n-i) \qquad (2.1.3.3.2.6.3.1\text{–}4)$$

then:

$$b'_L(n) = \sum_{i=0}^{\left\lfloor \frac{n}{L} \right\rfloor} z_L(n-iL) \qquad (2.1.3.3.2.6.3.1\text{–}5)$$

where $\lfloor x \rfloor$ is the floor function of x which evaluates to the largest integer $\le x$ $z_L(n)$ can be computed from $z_{L-1}(n)$ in a very efficient manner using (2.1.3.3.2.6.3.1–6) and (2.1.3.3.2.6.3.1–7).

$$z_L(n) = z_{L-1}(n-1) + r(-L)h(n) \qquad (2.1.3.3.2.6.3.1\text{–}6)$$

for $1 \le n \le N-1$ and

$$z_L(0) = r(-L)h(0) \qquad (2.1.3.3.2.6.3.1\text{–}7)$$

A further computational reduction can be achieved if a truncated impulse response is used in the computations associated with the long term lag determination process. The truncated impulse response can be written as:

$$h'(n) = \begin{cases} h(n) & , 0 \le n \le N_T - 1 \\ 0 & , N_T \le n \end{cases} \qquad (2.1.3.3.2.6.3.1\text{–}8)$$

A value of 21 for N_T, corresponding to a truncated impulse response 21 samples long, may be used.

Replacing h(n) with h'(n) in equations 2.1.3.3.2.6.3.1-4 and 2.1.3.3.2.6.3.1-6 results in a number of zero products that can be eliminated from the equations. These equations can be replaced by the following set of equations:

$$z_{L_{min}}(n) = \sum_{i=\max(0,\, n - N_T + 1)}^{\min(n,\, L_{min} - 1)} r(i-L_{min})\, h(n-i) \qquad (2.1.3.3.2.6.3.1\text{–}9)$$

$$z_L(n) = \begin{cases} z_{L-1}(n-1) + r(-L)h(n) & , 1 \le n \le N_T - 1 \\ z_{L-1}(n-1) & , N_T \le n \le N - 1 \end{cases}$$

(2.1.3.3.2.6.3.1–10)

Equation 2.1.3.3.2.6.3.1–9 is computed only for the minimum lag value and the array $z_L(n)$ for all other values of L is computed using 2.1.3.3.2.6.3.1–10.

Further computational reduction is possible when L > N–1 since then equation 2.1.3.3.2.6.3.1–5 reduces to

$$b'_L(n) = z_L(n) \qquad (2.1.3.3.2.6.3.1\text{–}11)$$

and computation of the energy term, G_L, in equation 2.1.3.3.2.6.3.1–1 simplifies to

$$G_L = E_L + \sum_{n=0}^{N_T-1} z_L^2(n) \qquad (2.1.3.3.2.6.3.1\text{–}12)$$

where

$$E_L = E_{L-1} + z_{L-1}^2(N_T - 1) - z_{L-1}^2(N-1) \qquad (2.1.3.3.2.6.3.1\text{–}13)$$

and

$$E_N = \sum_{n=N_T}^{N-1} z_N^2(n) \qquad (2.1.3.3.2.6.3.1\text{–}14)$$

These equations are used to determine G_L and C_L and the determination of the long term predictor lag, L, is based on maximizing equation 2.1.3.3.2.6.3.1–3. Once the lag is determined, the untruncated impulse response, h(n), is used to compute the weighted long term lag vector, $b'_L(n)$. This is done by computing $b'_L(n)$ as the zero state response of H(z) to $b_L(n)$ where $b_L(n)$ is:

$$b_L(n) = r(n - \lfloor (n+L)/L \rfloor L)\quad 0 \le n \le 39 \qquad (2.1.3.3.2.6.3.1\text{–}15)$$

2.1.3.3.2.6.3.2 Encoding of Lag

The lag, L, for each of the four subframes can take on the value of 20 to 146. This corresponds to 127 possible codes. Seven bits are used to encode each lag. The 128th code is used to indicate that the pitch predictor for that subframe is deactivated. The lag value is converted to the lag code as follows:

$$LAG\_x = L - 19 \qquad \text{if predictor not deactivated}$$

$$LAG\_x = 0 \qquad \text{if predictor deactivated}$$

where x is the numerals 1 through 4 for the 4 subframes. (2.1.3.3.2.6.3.2–1)

2.1.3.3.2.6.4 Codebook Excitation

The VSELP codec uses two excitation codebooks each of 2^M code vectors which are constructed from two sets of M basis vectors, where M = 7. Defining $v_{k,m}(n)$ as the m^{th} basis vector of the k^{th} codebook and $u_{k,i}(n)$ as the i^{th} code vector in the k^{th} codebook, then:

$$u_{k,i}(n) = \sum_{m=1}^{M} \theta_{im} \, v_{k,m}(n) \qquad (2.1.3.3.2.6.4\text{–}1)$$

where k = 1 for the first codebook and k = 2 for the second codebook and where $0 \le i \le 2^M - 1$; $0 \le n \le N-1$.

In other words, each code vector in the codebook is constructed as a linear combination of the M basis vectors. The linear combinations are defined by the θ parameters. θ_{im} is defined as:

$\theta_{im} = +1$ if bit m of codeword i = 1

$\theta_{im} = -1$ if bit m of codeword i = 0.

The codebook construction for the VSELP codec can restated as follows. Code vector i is constructed as the sum of the M basis vectors where the sign (plus or minus) of each basis vector is determined by the state of the corresponding bit in codeword i. Note that if we complement all the bits in codeword i, the corresponding code vector is the negative of code vector i. Therefore, for every code vector, its negative is also a code vector in the codebook. These pairs are called complementary code vectors since the corresponding codewords are complements of each other. The basis vectors ($v_{k,m}(n)$) which specify the two VSELP codebooks are given in Table 2.1.3.3.2.6.4–1.

Table 2.1.3.3.2.6.4–1 Format in Which the Basis Vector Samples Are Given:

1	2	3	4
5	6	7	8
.	.	.	.
.	.	.	.
.	.	.	.
33	34	35	36
37	38	39	40

BASIS VECTOR # 1 FROM CODEBOOK # 1

0.3451242149E+00	-0.1908946037E+00	-0.1863791049E+00	-0.2809082866E+00
-0.5949888006E-01	-0.2898133993E+00	-0.2007001042E+00	-0.2624697983E+00
-0.2319182009E+00	-0.1809999943E+00	0.3789095208E-01	-0.2147784978E+00
0.1241798997E+00	-0.1218566000E+00	0.2359705418E-01	0.2713254094E+00
0.1632671058E+00	-0.4970406741E-01	0.1902845949E+00	-0.1335213892E-01
0.9223834425E-01	-0.5935087055E-01	0.2032784373E-01	-0.2071827054E+00
0.2095990814E-01	-0.1549381018E+00	-0.2140610069E+00	-0.2329927981E+00
-0.3249342367E-01	-0.1220064014E+00	0.1809466928E+00	0.6742917746E-01
0.1195665970E+00	0.3998436928E+00	0.2750256956E+00	0.1396846026E+00
-0.2264889032E+00	0.2459658980E+00	-0.1478482038E+00	0.1349629015E+00

BASIS VECTOR # 2 FROM CODEBOOK # 1

-0.1649273038E+00	-0.1280633956E+00	0.2084657997E+00	0.2897260711E-01
-0.1229320988E+00	0.9742700309E-01	0.5553475767E-01	0.3294681013E+00
-0.1346358955E+00	0.1256469041E+00	-0.1168436036E+00	0.6231806241E-02
0.7953327894E-01	0.3193782866E+00	-0.2746013105E+00	-0.1363205444E-01
-0.1766237020E+00	-0.1918185949E+00	-0.1746349931E+00	-0.2069617957E+00
-0.1570205986E+00	-0.2679320872E+00	-0.2100628428E-01	-0.2031996995E+00
-0.2044696063E+00	-0.4844427574E-02	-0.1820494980E+00	-0.7023712248E-01
-0.5200922117E-01	-0.1236483976E+00	0.1006868035E+00	-0.4063146114E+00
0.1314809024E+00	-0.1702741603E-02	0.3849417865E+00	0.3761824071E+00
0.1832554936E+00	0.2255230993E+00	0.3952260017E+00	0.1965305060E+00

1 BASIS VECTOR # 3 FROM CODEBOOK # 1

0.1222262010E+00	-0.2443725429E-01	0.9521071613E-01	0.2051250041E+00
-0.2647739053E+00	-0.4311279953E-01	-0.2744256854E+00	0.4825744405E-01
-0.1441915929E+00	0.1775548011E+00	0.2227558941E+00	0.4199514985E+00
0.3407743871E+00	0.3164913058E+00	0.1395058036E+00	-0.2050642073E+00
0.6457627565E-01	-0.3537886143E+00	-0.5985410511E-01	-0.2940579951E+00
0.1644386053E+00	-0.9712598473E-01	0.2345167994E+00	-0.1141005009E+00
0.3247086108E+00	-0.7059235126E-01	-0.1467435956E+00	-0.1093520969E+00
0.1290429980E+00	-0.3067413867E+00	0.6985686719E-01	0.3380126059E+00
0.1060701013E+00	0.4212953150E-02	0.2529056966E+00	-0.5316415336E-02
-0.9389756620E-01	-0.1408378035E+00	-0.1945132017E+00	-0.3807552159E-01

2 BASIS VECTOR # 4 FROM CODEBOOK # 1

-0.2002453059E+00	0.3095070124E+00	0.1193293035E+00	0.2501963079E+00
0.7344523072E-01	0.2895031869E+00	0.4940271005E-01	-0.1672842950E+00
0.1761247963E+00	0.7341763377E-01	-0.1781246960E+00	0.4386122897E-01
-0.4659612477E-01	-0.7253613323E-01	-0.2906739116E+00	-0.6229455397E-01
-0.1309871972E+00	0.7658090442E-01	0.1743921936E+00	0.1855054945E+00
0.3523846865E+00	0.1146479025E+00	0.1589893997E+00	0.2222200036E+00
-0.1127394941E-01	-0.2681927979E+00	-0.3662436008E+00	-0.3717831075E+00
0.2991584130E-01	-0.3533985913E+00	-0.4226866737E-01	0.6729383767E-01
-0.3311626986E-01	0.3588928878E+00	-0.2317036986E+00	0.1110844016E+00
-0.1048915014E+00	0.6802166253E-01	0.3553674743E-01	0.2772972547E-01

3 BASIS VECTOR # 5 FROM CODEBOOK # 1

-0.6483641267E+00	-0.1471579727E-01	-0.3926123083E+00	-0.3888195008E-01
0.2079516463E-01	0.1957547069E+00	-0.9889058024E-01	0.3825030029E+00
0.8439254016E-01	0.3003462851E+00	0.1768756062E+00	0.2268030941E+00
0.3840109110E+00	-0.1217688024E+00	0.1454651952E+00	-0.5188317969E-01
0.1141197011E+00	-0.2478182875E-01	0.9596040845E-01	0.1432708949E+00
-0.5413892493E-01	0.3947615623E-02	-0.2586893141E+00	-0.2089742571E-01
-0.2101484984E+00	-0.6067659706E-01	-0.2703967504E-01	0.6947388500E-01
0.1313298047E+00	0.6357792765E-01	0.2391532250E-01	0.6551270187E-01
0.5597909912E-01	-0.4799355194E-01	-0.6142149866E-01	-0.3112792969E+00
-0.3918431103E+00	-0.8220961690E-01	0.2932965523E-02	-0.1451980025E+00

1 BASIS VECTOR # 6 FROM CODEBOOK # 1

0.3414244056E+00	0.2305835932E+00	-0.1154242009E+00	0.4631759599E-01
-0.6643511355E-01	0.8361395448E-01	-0.9363006055E-01	-0.2741353214E-01
0.7112574577E-01	-0.1254739016E+00	0.7751072943E-01	0.1796556264E-01
0.5026462078E+00	0.7896416634E-01	0.4279431999E+00	0.2176973969E+00
0.2289295048E+00	0.4618147761E-01	0.1656796932E+00	-0.5426255241E-01
0.1128297970E+00	-0.3721603751E-01	0.7740236074E-01	-0.2443609983E+00
-0.1894937009E+00	0.1477535069E+00	-0.5043362975E+00	0.2714478597E-01
-0.1350484043E+00	-0.3649033234E-01	-0.1041088998E+00	-0.2749828100E+00
-0.5220643431E-01	-0.3343094885E+00	-0.3461129069E+00	-0.3607497364E-01
0.8886242658E-01	-0.2545958944E-01	0.4283397645E-01	-0.6496996619E-02

2 BASIS VECTOR # 7 FROM CODEBOOK # 1

-0.9914957881E+00	-0.4214186221E-01	-0.2065369934E+00	-0.6139709428E-01
-0.2154099382E-01	0.1004114971E+00	0.1811787933E+00	0.5004148930E-01
0.2562974095E+00	-0.1523203999E+00	-0.2674589306E-01	-0.1296720058E+00
0.4938440397E-01	0.1061969027E+00	0.1991406977E+00	0.2695614994E+00
-0.2002000436E-01	0.2905817032E+00	-0.3127651755E-02	0.1016449034E+00
0.1299401969E+00	0.2303000987E+00	0.2902364135E+00	0.1896886975E+00
0.3253236115E+00	0.2585191838E-01	-0.2631102223E-02	-0.7936273515E-01
-0.9870006144E-01	-0.1658474952E+00	-0.1042916998E+00	-0.1593011022E+00
0.7002063841E-01	0.9161889553E-02	0.5557846278E-02	-0.2852319553E-01
0.4387573805E-02	-0.7425554097E-01	-0.1108988002E+00	-0.1056471020E+00

3 BASIS VECTOR # 1 FROM CODEBOOK # 2

-0.1078469992E+01	0.3220184147E+00	0.3572095037E+00	0.6032103896E+00
0.3291260004E+00	-0.2068174034E+00	-0.7850720733E-01	0.7766201347E-01
-0.5539268255E+00	-0.4499691129E+00	-0.2874793112E+00	0.1191606000E+00
-0.1273895055E+00	0.2862497866E+00	0.2214280069E+00	0.2592194974E+00
-0.1962423027E+00	-0.5959221721E-01	0.1745598018E+00	-0.8334506303E-02
-0.1402842999E+00	-0.2419897020E+00	0.9168544412E-01	-0.2495622039E+00
-0.2419583052E+00	0.1644829959E+00	-0.2559731901E+00	0.5646622181E+00
-0.2104290761E-01	0.9073065221E-01	-0.3278034925E+00	0.2478367984E+00
0.2951978147E+00	0.2485664934E+00	-0.1241953000E+00	-0.5802686140E-01
-0.3626115024E+00	-0.3764331713E-01	-0.8315514028E-01	-0.1295665950E+00

1 BASIS VECTOR # 2 FROM CODEBOOK # 2

-0.6171314716E+00	0.1042459980E+00	0.1904276013E+00	-0.2003915012E+00
0.2364176959E+00	-0.3931302130E+00	-0.7451597601E-01	-0.1113720983E+00
-0.3296768069E+00	0.5505855009E-01	-0.2221453041E+00	0.6063252091E+00
0.1949710995E+00	-0.9999454767E-01	-0.2479213029E+00	-0.1112871021E+00
0.7108733803E-01	-0.3127546236E-01	-0.3499733284E-02	0.4116210938E+00
0.6289582253E+00	-0.1262892317E-01	0.3541105092E+00	-0.2191990018E+00
-0.4144948125E+00	0.3219327331E-01	0.1044111997E+00	0.1266452968E+00
0.4700252116E+00	0.2069593072E+00	0.2521953881E+00	0.5423704535E-01
-0.1685699970E+00	-0.5957524776E+00	0.1484079063E+00	0.3758538067E+00
0.3699175119E+00	0.2625274956E+00	-0.2196810991E+00	0.2226922959E+00

2 BASIS VECTOR # 3 FROM CODEBOOK # 2

-0.7037442923E+00	0.1780209988E+00	0.1386262029E+00	0.1773904264E-01
0.3194114864E+00	-0.8156231046E+00	-0.3730263934E-01	0.1182738021E+00
-0.4221254960E-01	0.1008443981E+00	0.3434633017E+00	-0.3526256084E+00
-0.2351640016E+00	-0.2203640044E+00	-0.3177471692E-02	0.1706431061E+00
0.7822614908E-01	-0.5083335042E+00	-0.2581757903E+00	-0.5118042827E+00
0.2634621859E+00	0.3051618934E+00	0.1222215965E+00	-0.1114903986E+00
-0.3546279967E+00	-0.8888566494E-01	-0.1979334950E+00	-0.4040746093E+00
0.1048382279E-01	0.6411200762E-01	0.2749992907E+00	0.3400909901E+00
-0.2861019596E-01	-0.3443039060E+00	-0.4039180875E+00	-0.3798497021E+00
0.3202857077E+00	-0.2101213932E+00	0.4261999130E+00	0.3337397873E+00

3 BASIS VECTOR # 4 FROM CODEBOOK # 2

0.7885073125E-01	0.1641726047E+00	0.2769562006E+00	0.3139050007E+00
-0.1808016002E+00	-0.6696839929E+00	0.2347880006E+00	0.4427241981E+00
0.3996557891E+00	0.5498163104E+00	-0.1858830005E+00	-0.2986466885E+00
0.9812835604E-01	-0.4205571115E+00	0.4010680914E+00	0.3347500414E-01
0.2540239990E+00	0.4309037924E+00	0.1113025993E+00	0.2318263054E+00
-0.2789230049E+00	-0.1045858022E-01	0.1381077021E+00	-0.4480378926E+00
0.8657258004E-01	-0.2983936071E+00	0.1923187971E+00	0.2231222987E+00
0.6417953223E-01	-0.1814893037E+00	0.2403858006E+00	0.3521871194E-01
0.2641254067E+00	0.2590672076E+00	-0.1733974181E-01	0.1440864950E+00
0.5846006870E+00	-0.1969480067E+00	-0.2235774994E+00	-0.3034487963E+00

1 BASIS VECTOR # 5 FROM CODEBOOK # 2

0.4358376190E-01	-0.5805857107E-01	0.4501081109E+00	-0.2082135975E+00
-0.3582383990E+00	-0.2613596022E+00	-0.6745511889E+00	0.5382819176E+00
-0.1477800962E-01	-0.5667730793E-01	0.1898874938E+00	0.3535073102E+00
0.4257721826E-01	-0.2010733038E+00	-0.3887566030E+00	-0.3210910857E+00
0.7393498719E-01	0.4713085890E+00	-0.7108450681E-01	-0.1935663968E+00
0.1660837978E+00	-0.3511838615E-01	0.1417755932E+00	-0.2500975132E+00
-0.1513005942E+00	0.1157625020E+00	0.4493466914E+00	0.2846056968E-01
-0.2096074969E+00	-0.3893598020E+00	-0.2374109030E+00	-0.6184939742E+00
-0.3851557076E+00	0.4619333148E+00	0.9302373976E-01	-0.3500598073E+00
0.1615410298E-01	-0.1559370011E+00	-0.1810939014E+00	-0.1666457951E+00

2 BASIS VECTOR # 6 FROM CODEBOOK # 2

0.6736537218E+00	0.1836300045E+00	-0.1169203967E+00	-0.2385098040E+00
-0.3071058728E-01	0.3613345921E+00	0.4077064991E+00	0.2359752059E+00
0.4120293260E-01	-0.1880857050E+00	-0.1632833034E+00	-0.6836611032E-01
-0.4735979810E-01	-0.3098683432E-01	0.1077236980E+00	0.2860186100E+00
-0.3181805015E+00	0.2275553942E+00	-0.5157095194E+00	0.1487997267E-01
0.1960210055E+00	0.5612939000E+00	-0.2136532962E+00	-0.1722970009E+00
-0.3482623994E+00	-0.2468501031E+00	-0.2466483414E-01	-0.2039584070E+00
0.5595381260E+00	-0.1073065028E+00	-0.3460974619E-01	0.5144857243E-01
0.1268441975E+00	-0.2642670870E+00	0.6754587889E+00	0.1380926967E+00
-0.4250771999E+00	-0.6028810143E+00	-0.1822202951E+00	-0.2909131944E+00

3 BASIS VECTOR # 7 FROM CODEBOOK # 2

-0.6942132115E+00	0.1113955006E+00	0.6013740897E+00	0.1951819062E+00
0.5133929253E+00	0.1132920980E+00	0.1437312961E+00	-0.2392677963E+00
-0.4471820891E+00	0.3258281052E+00	0.4280638099E+00	0.1897481978E+00
-0.1792849004E+00	-0.2804139853E+00	-0.4632335901E-01	0.5034636855E+00
0.1647147983E+00	0.1746135056E+00	0.2235112935E+00	-0.4559667110E+00
0.3063598871E+00	-0.4095937014E+00	-0.2687213123E+00	-0.3071638942E+00
0.2464583963E+00	0.1107138991E+00	0.5172047019E-01	-0.1779457927E+00
-0.4489682913E+00	0.4342001975E+00	-0.3757492900E+00	0.7282961905E-01
-0.4642477930E+00	0.8351081610E-01	0.2392690927E+00	0.3020069003E+00
0.1004394703E-01	-0.3548842967E+00	-0.2936468124E+00	-0.9987259656E-01

The excitation Codebook Search procedure takes place after the long term predictor lag, L, has been determined. The Codebook Search procedure sequentially chooses one code vector from the first VSELP codebook and then chooses one code vector from the second VSELP codebook. Define:

I = codeword selected from first VSELP codebook

H = codeword selected from the second VSELP codebook.

2.1.3.3.2.6.4.1 Filtering of Basis Vectors

To perform the codebook searches, the zero state response of each basis vector to H(z) must be computed for both codebooks. Define $q_{k,m}(n)$ to be the zero state response of H(z) to basis vector $v_{k,m}(n)$; $0 \leq n \leq N-1$. From the definition of the VSELP codebook (Table 2.1.3.3.2.6.4–1), the zero state response of each code vector $f_{k,i}(n)$, can be expressed as:

$$f_{k,i}(n) = \sum_{m=1}^{M} \theta_{im} \, q_{k,m}(n) \qquad (2.1.3.3.2.6.4.1\text{–}1)$$

2.1.3.3.2.6.4.2 Orthogonalization of Filtered Basis Vectors

The selection of the code vector from the first codebook must account for the previous selection of the long term predictor lag, L. The selection of the code vector from the second codebook must account for the selection of both the long term predictor lag, L, and the codeword selected from the first codebook, I. One technique which may be employed uses orthogonalization procedures to decouple the selection process for the codebook excitation vectors from previously determined excitation components.

2.1.3.3.2.6.4.2.1 Orthogonalization for Codebook 1

Prior to the first codebook search each filtered basis vector for the first codebook, $q_{1,m}(n)$, may be made orthogonal to $b'_L(n)$, the zero state response of H(z) to the long term prediction vector $b_L(n)$.

Defining:

$$\Gamma = \sum_{n=0}^{N-1} \left(b'_L(n)\right)^2 \qquad (2.1.3.3.2.6.4.2.1\text{–}1)$$

and

$$\Psi_m = \sum_{n=0}^{N-1} b'_L(n) \, q_{1,m}(n) \qquad (2.1.3.3.2.6.4.2.1\text{–}2)$$

for $1 \leq m \leq M$; then $q'_{1,m}(n)$, the orthogonalized filtered basis vectors, can be computed by:

$$q'_{1,m}(n) = q_{1,m}(n) - \left(\frac{\Psi_m}{\Gamma}\right) b'_L(n) \qquad (2.1.3.3.2.6.4.2.1\text{–}3)$$

for $1 \leq m \leq M$ and $0 \leq n \leq N-1$.

The orthogonalized filtered code vectors can now be expressed as:

$$f'_{1,i}(n) = \sum_{m=1}^{M} \theta_{im}\, q'_{1,m}(n) \qquad (2.1.3.3.2.6.4.2.1\text{–}4)$$

for $0 \le i \le 2^M - 1$ and $0 \le n \le N-1$.

Using the orthogonalized filtered codebook vectors, $f'_{1,i}(n)$, the expression to be minimized is:

$$E'_{1,i} = \sum_{n=0}^{N-1} (p(n) - \gamma'_1 f'_{1,i}(n))^2 \qquad (2.1.3.3.2.6.4.2.1\text{–}5)$$

where γ'_1 is optimized for each code vector i.

2.1.3.3.2.6.4.2.2 Orthogonalization for Codebook 2

The filtered basis vectors for the second codebook may be orthogonalized to both $b'_L(n)$ and $f_{1,I}(n)$. This can be done by first orthogonalizing the $q_{2,m}(n)$ vectors with respect to $b'_L(n)$ in the same manner as for the first codebook. The resulting vectors are then orthogonalized with respect to $f'_{1,I}(n)$ using a similar procedure. Since $f'_{1,I}(n)$ is orthogonal to $b'_L(n)$, the resulting vectors, $q'_{2,m}(n)$, will be orthogonal to both $b'_L(n)$ and $f_{1,I}(n)$ (this is an implementation of Gram-Schmidt orthogonalization).

The orthogonalized filtered code vectors for the second codebook can now be expressed as:

$$f_{2,i}(n) = \sum_{m=1}^{M} \theta_{im}\, q'_{2,m}(n) \qquad (2.1.3.3.2.6.4.2.2\text{–}1)$$

for $0 \le i \le 2^M - 1$ and $0 \le n \le N-1$.

Using the decorrelated filtered codebook vectors, $f'_{2,i}(n)$, the expression to be minimized for the second codebook search is:

$$E'_{2,i} = \sum_{n=0}^{N-1} (p(n) - \gamma'_2 f'_{2,i}(n))^2 \qquad (2.1.3.3.2.6.4.2.2\text{–}2)$$

2.1.3.3.2.6.4.3 VSELP Codebook Search

The Codebook Search procedure should find the codeword i which minimizes:

$$E'_{k,i} = \sum_{n=0}^{N-1} (p(n) - \gamma'_k f'_{k,i}(n))^2 \qquad (2.1.3.3.2.6.4.3\text{–}1)$$

where k =1 for the first codebook and k = 2 for the second codebook and where γ'_k is optimal for each code vector i. In the rest of this section the subscript k indicating the first or second codebook will be dropped. Once we have the filtered and orthogonalized basis vectors, the actual Codebook Search procedures are identical.

Defining :

$$C_i = \sum_{n=0}^{N-1} f'_i(n)\, p(n) \qquad (2.1.3.3.2.6.4.3\text{–}2)$$

and

$$G_i = \sum_{n=0}^{N-1} (f'_i(n))^2 \qquad (2.1.3.3.2.6.4.3\text{–}3)$$

then the code vector shall be chosen as the one which maximizes :

$$\frac{(C_i)^2}{G_i} \qquad (2.1.3.3.2.6.4.3\text{–}4)$$

The search process should evaluate (2.1.3.3.2.6.4.3–4) for each code vector. The code vector which maximizes (2.1.3.3.2.6.4.3–4) is then chosen. Using properties of the VSELP codebook construction, the computations required for computing C_i and G_i can be greatly simplified.

Defining:

$$R_m = 2\sum_{n=0}^{N-1} q'_m(n)\, p(n) \qquad (2.1.3.3.2.6.4.3\text{–}5)$$

for $1 \le m \le M$ and

$$D_{mj} = 4\sum_{n=0}^{N-1} q'_m(n)\, q'_j(n) \qquad (2.1.3.3.2.6.4.3\text{–}6)$$

for $1 \le m \le j \le M$

C_i can be expressed as:

$$C_i = \frac{1}{2}\sum_{m=1}^{M} \theta_{im} R_m \qquad (2.1.3.3.2.6.4.3\text{–}7)$$

and G_i can be expressed as:

$$G_i = \frac{1}{2}\sum_{j=2}^{M} \sum_{m=1}^{j-1} \theta_{im}\, \theta_{ij}\, D_{mj} + \frac{1}{4}\sum_{j=1}^{M} D_{jj} \qquad (2.1.3.3.2.6.4.3\text{–}8)$$

Assuming that codeword u differs from codeword i in only one bit position, say position v such that $\theta_{uv} = -\theta_{iv}$ and $\theta_{um} = \theta_{im}$ for $m \neq v$ then:

$$C_u = C_i + \theta_{uv} R_v \qquad (2.1.3.3.2.6.4.3\text{–}9)$$

and

$$G_u = G_i + \sum_{j=1}^{v-1} \theta_{uj}\, \theta_{uv}\, D_{jv} + \sum_{j=v+1}^{M} \theta_{uj}\, \theta_{uv}\, D_{vj} \qquad (2.1.3.3.2.6.4.3\text{–}10)$$

If the codebook search is structured such that each successive codeword evaluated differs from the previous codeword in only one bit position, then (2.1.3.3.2.6.4.3–9) and (2.1.3.3.2.6.4.3–10) can be used to update C_i and G_i in a very efficient manner. Sequencing of the codewords in this manner is accomplished using a binary Gray code.

Note that complementary codewords (see Section 2.1.3.3.2.6.4) will have equivalent values for (2.1.3.3.2.6.4.3–4). Therefore only half of the code vectors need to be evaluated. Once the code vector which maximizes (2.1.3.3.2.6.4.3–4) is found, the sign of C_i for that code vector will determine whether that code vector or its complement will yield a positive gain. If C_i is positive then i is the selected codeword, if C_i is negative the ones complement of i is selected as the codeword.

2.1.3.3.2.6.4.4 Encoding of Excitation Codewords

The code value for the first codebook, CODE1_x, is the codeword I as derived by the Codebook Search procedure in Section 2.1.3.3.2.6.4. The least significant bit corresponds to the first basis vector for the codebook.

The code value for the second codebook, CODE2_x, is the codeword H as derived by the Codebook Search procedure in Section 2.1.3.3.2.6.4. The least significant bit corresponds to the first basis vector for the codebook.

2.1.3.3.2.6.5 Quantization of Gains

The weighted error per sample in a subframe is given by

$$e(n) = p(n) - \beta\, c'_0(n) - \gamma_1\, c'_1(n) - \gamma_2\, c'_2(n) \quad 0 \le n \le N-1$$

(2.1.3.3.2.6.5–1)

where $p(n)$ is the weighted input to be matched, minus zero input response of $H(z)$

$c'_0(n)$ is the weighted long term prediction vector – $b'_L(n)$

$c'_1(n)$ is the weighted code vector selected from codebook 1– $f_{1,I}(n)$

$c'_2(n)$ is the weighted code vector selected from codebook 2– $f_{2,H}(n)$

β is the long term predictor coefficient

γ_1 is the gain scaling the code vector from codebook 1

γ_2 s the gain scaling the code vector from codebook 2

Consequently the total weighted error squared for a subframe is given by:

$$E = \sum_{n=0}^{N-1} e^2(n) = \sum_{n=0}^{N-1} (p(n) - \beta\, c'_0(n) - \gamma_1 c'_1(n) - \gamma_2 c'_2(n))^2$$

(2.1.3.3.2.6.5–2)

To simplify the error equation, E may be expressed in terms of correlations among vectors $p(n)$, $c'_0(n)$, $c'_1(n)$, and $c'_2(n)$.

Let

$$R_{pp} = \sum_{n=0}^{N-1} p(n)p(n)$$

(2.1.3.3.2.6.5–3)

$$R_{pc}(k) = \sum_{n=0}^{N-1} p(n)c'_k(n) \qquad k = 0, 2$$

(2.1.3.3.2.6.5–4)

$$R_{cc}(k,j) = \sum_{n=0}^{N-1} c'_k(n)\, c'_j(n) \qquad k = 0, 2; \quad j = k, 2$$

(2.1.3.3.2.6.5–5)

$$R_{cc}(k,j) = R_{cc}(j,k)$$

(2.1.3.3.2.6.5–6)

Incorporating the correlations into the error expression yields

$$E = R_{pp} - 2\,\beta R_{pc}(0) - 2\sum_{j=1}^{2} \gamma_j R_{pc}(j) + 2\beta\sum_{j=1}^{2} \gamma_j R_{cc}(0,j) + 2\gamma_1\gamma_2 R_{cc}(1,2)$$
$$+ \beta^2 R_{cc}(0,0) + \sum_{j=1}^{2} \gamma_j^2 R_{cc}(j,j)$$

(2.1.3.3.2.6.5–7)

Minimizing the weighted error consists of jointly optimizing β, the long term predictor coefficient, with γ_1 and γ_2, the gain terms to minimize 2.1.3.3.2.6.5–7.

2.1.3.3.2.6.5.1 Transformation of Gains to GS, P0 and P1

Define ex(n) to be the excitation function at a given subframe. ex(n) is a linear combination of the long term predictor vector scaled by β, the long term predictor coefficient, and of the code vectors scaled by γ_1 and γ_2, their respective gains.

$$ex(n) = \beta\, c_0(n) + \gamma_1\, c_1(n) + \gamma_2\, c_2(n) \qquad 0 \le n \le N-1 \qquad (2.1.3.3.2.6.5.1\text{–}1)$$

where $c_0(n)$ is the unweighted long term prediction vector, $b_L(n)$
$c_1(n)$ is the unweighted code vector selected from codebook 1, $u_{1,I}(n)$
$c_2(n)$ is the unweighted code vector selected from codebook 2, $u_{2,H}(n)$

The energy in each excitation vector is given by

$$R_x(k) = \sum_{n=0}^{N-1} c^2_k(n) \qquad k = 0, 2 \qquad (2.1.3.3.2.6.5.1\text{–}2)$$

Let RS be the approximate residual energy at a given subframe. RS is a function of N, R'q(0), and of the normalized prediction gain of the LPC filter.

$$RS = N\, R'_q(0) \prod_{i=1}^{N_p} (1 - r_i^2) \qquad (2.1.3.3.2.6.5.1\text{–}3)$$

where r_i is the i^{th} reflection coefficient for the subframe corresponding to the set of direct form filter coefficients (α_i's) for the subframe.

GS, the energy offset parameter, is a coded parameter which adjusts the estimated value of RS. Define:

$$R = GS\ RS \qquad (2.1.3.3.2.6.5.1\text{–}4)$$

Define P0, the energy contribution of the long term prediction vector as a fraction of the total excitation energy at a subframe, as

$$P0 = \frac{\beta^2 R_x(0)}{R} \quad \text{where } 0 \le P0 \le 1 \qquad (2.1.3.3.2.6.5.1\text{–}5)$$

Similarly, P1, the energy contribution of the code vector selected from the first codebook as a fraction of the total excitation energy at a subframe, is defined as:

$$P1 = \frac{\gamma_1^2 R_x(1)}{R} \quad \text{where } P0 + P1 \le 1 \qquad (2.1.3.3.2.6.5.1\text{–}6)$$

Thus β, γ_1, and γ_2 are replaced by three new parameters: P0, P1, and GS. The transformations relating β, γ_1, and γ_2 to GS, P0, and P1 are given by

$$\beta = \sqrt{\frac{RS\ GS\ P0}{R_x(0)}} \qquad (2.1.3.3.2.6.5.1\text{–}7)$$

$$\gamma_1 = \sqrt{\frac{RS\ GS\ P1}{R_x(1)}} \qquad (2.1.3.3.2.6.5.1\text{–}8)$$

$$\gamma_2 = \sqrt{\frac{RS\ GS\ (1-P0-P1)}{R_x(2)}} \qquad (2.1.3.3.2.6.5.1\text{–}9)$$

2.1.3.3.2.6.5.2 Vector Quantization and Encoding of GS, P0 and P1

Replacing the β, γ_1, and γ_2 in (2.1.3.3.2.6.5–7) by the equivalent expressions in terms of GS, P0, P1, and $R_x(k)$ results in the updated weighted error equation

$$\begin{aligned} E = R_{pp} &- a\sqrt{GS\ P0} - b\sqrt{GS\ P1} - c\sqrt{GS\ (1-P0-P1)} + d\ GS\sqrt{P0\ P1} \\ &+ e\ GS\sqrt{P0\ (1-P0-P1)} + f\ GS\sqrt{P1\ (1-P0-P1)} \\ &+ g\ GS\ P0 + h\ GS\ P1 + i\ GS\ (1-P0-P1) \end{aligned}$$

(2.1.3.3.2.6.5.2–1)

where

$$a = 2R_{pc}(0)\sqrt{\frac{RS}{R_x(0)}} \qquad (2.1.3.3.2.6.5.2\text{–}2)$$

$$b = 2R_{pc}(1)\sqrt{\frac{RS}{R_x(1)}} \qquad (2.1.3.3.2.6.5.2\text{–}3)$$

$$c = 2R_{pc}(2)\sqrt{\frac{RS}{R_x(2)}} \qquad (2.1.3.3.2.6.5.2\text{–}4)$$

$$d = \frac{2R_{cc}(0,1)\,RS}{\sqrt{R_x(0)R_x(1)}} \qquad (2.1.3.3.2.6.5.2\text{–}5)$$

$$e = \frac{2R_{cc}(0,2)\,RS}{\sqrt{R_x(0)R_x(2)}} \qquad (2.1.3.3.2.6.5.2\text{–}6)$$

$$f = \frac{2R_{cc}(1,2)\,RS}{\sqrt{R_x(1)R_x(2)}} \qquad (2.1.3.3.2.6.5.2\text{–}7)$$

$$g = \frac{R_{cc}(0,0)\,RS}{R_x(0)} \qquad (2.1.3.3.2.6.5.2\text{–}8)$$

$$h = \frac{R_{cc}(1,1)\,RS}{R_x(1)} \qquad (2.1.3.3.2.6.5.2\text{–}9)$$

$$i = \frac{R_{cc}(2,2)\,RS}{R_x(2)} \qquad (2.1.3.3.2.6.5.2\text{–}10)$$

GS, P0 and P1 are vector quantized. The first step in quantizing vector {GS, P0, P1} consists of calculating the parameters required by the error equation:

$R_{cc}(k,j)$ $k = 0, 2;\ j = k, 2$

$R_x(0), R_x(1), R_x(2)$

RS

$R_{pc}(k)$ $k = 0, 2$

a, b, c, d, e, f, g, h, i

Next (Table 2.1.3.3.2.6.5.2–1) is evaluated for each vector in the {GS, P0, P1}codebook, and the vector which minimizes the weighted error is selected. The {GS, P0, P1} codebook is given in Table 2.1.3.3.2.6.5.2–1.

Table 2.1.3.3.2.6.5.2–1 GS, P0, P1 Codebook

INDEX #	GS	P0	P1
0	0.1674511004E-02	0.1628413945E+00	0.5090150237E+00
1	0.1334269880E-02	0.5378435850E+00	0.2362515032E+00
2	0.3112269565E-02	0.1365114003E+00	0.7425394058E+00
3	0.3868540749E-02	0.3757469952E+00	0.5166236758E+00
4	0.3916129004E-02	0.3697459996E+00	0.1677096933E+00
5	0.2203964163E-02	0.6928331256E+00	0.2034170032E+00
6	0.6334181409E-02	0.5041198730E+00	0.2782799006E+00
7	0.6364202593E-02	0.7150015831E+00	0.9318676591E-01
8	0.6032689475E-02	0.1773163974E+00	0.4278650880E+00
9	0.8498853073E-02	0.2549907863E+00	0.1608013064E+00
10	0.1103511825E-01	0.2339199036E+00	0.4263555110E+00
11	0.1213169005E-01	0.4288598895E+00	0.1452523023E+00
12	0.1125547010E-01	0.5531352162E+00	0.2692165077E+00
13	0.1499444060E-01	0.6825100780E+00	0.1179910004E+00
14	0.1991050877E-01	0.5739204288E+00	0.2914825976E+00
15	0.2668176405E-01	0.7299969792E+00	0.1241495013E+00
16	0.1216377132E-01	0.2081580013E+00	0.6651288867E+00
17	0.1665616408E-01	0.2399802953E+00	0.5246518254E+00
18	0.2211280540E-01	0.5327664316E-01	0.5560309887E+00
19	0.4146175086E-01	0.4332519695E-01	0.3001616001E+00
20	0.1824236475E-01	0.4052976966E+00	0.2251251936E+00
21	0.2846666984E-01	0.2728624940E+00	0.7517775893E-01
22	0.4009734094E-01	0.2904489934E+00	0.1128304973E+00
23	0.3304062411E-01	0.5624005198E+00	0.1144611016E+00
24	0.3026418947E-01	0.4282687008E+00	0.2743416131E+00
25	0.4202774912E-01	0.4557394087E+00	0.2395865023E+00
26	0.3683714196E-01	0.6080287099E+00	0.2409482002E+00
27	0.4296181723E-01	0.7553868294E+00	0.1563764066E+00
28	0.5061166734E-01	0.5004783869E+00	0.1042283028E+00
29	0.6330294907E-01	0.6044433713E+00	0.9549445659E-01
30	0.5410422012E-01	0.8002007008E+00	0.5705973133E-01
31	0.7002255321E-01	0.8490940928E+00	0.8035162836E-01
32	0.2551996335E-01	0.2582708895E+00	0.5111098289E+00
33	0.3739386052E-01	0.1589131057E+00	0.5667265058E+00
34	0.3554347903E-01	0.1742120981E+00	0.3600822091E+00
35	0.4851049930E-01	0.2376495004E+00	0.3000344038E+00
36	0.6062155217E-01	0.5450420082E-01	0.4720115960E+00
37	0.1110351011E+00	0.4298216850E-01	0.2424717993E+00

38	0.7409280539E-01	0.2410012931E+00	0.3175866902E+00
39	0.1102432981E+00	0.2211526036E+00	0.1535796970E+00
40	0.4073541611E-01	0.3339963853E+00	0.5350126028E+00
41	0.5102888495E-01	0.3695037961E+00	0.3936882913E+00
42	0.5682564899E-01	0.3825046122E+00	0.2120109946E+00
43	0.7335599512E-01	0.4522989094E+00	0.2309546024E+00
44	0.5556463450E-01	0.5706608891E+00	0.2843337059E+00
45	0.6869171560E-01	0.6323106885E+00	0.2636449933E+00
46	0.8424820751E-01	0.6379644871E+00	0.1834726036E+00
47	0.9327419102E-01	0.7574281096E+00	0.1561726928E+00
48	0.8650545031E-01	0.4290732145E+00	0.1055454984E+00
49	0.9788852930E-01	0.4905169904E+00	0.1626531035E+00
50	0.1121798009E+00	0.5697429776E+00	0.6549628824E-01
51	0.1267466992E+00	0.6473875046E+00	0.7873713970E-01
52	0.9539756179E-01	0.5148016214E+00	0.2657338977E+00
53	0.1166627035E+00	0.5716524720E+00	0.2173192054E+00
54	0.1064478979E+00	0.6332104802E+00	0.2814091146E+00
55	0.1370353997E+00	0.7724838853E+00	0.1699541062E+00
56	0.7964644581E-01	0.6711853147E+00	0.9912785143E-01
57	0.9281945974E-01	0.7759063840E+00	0.6934114546E-01
58	0.1014161035E+00	0.8620666265E+00	0.3112772666E-01
59	0.1022674963E+00	0.9087637067E+00	0.5598694459E-01
60	0.1174440011E+00	0.7431836128E+00	0.1049233973E+00
61	0.1427710056E+00	0.7971041203E+00	0.9140700102E-01
62	0.1335725933E+00	0.8768588901E+00	0.4184054211E-01
63	0.1557984948E+00	0.9022567868E+00	0.6620954722E-01
64	0.6495805830E-01	0.6296302378E-01	0.7413570881E+00
65	0.9106755257E-01	0.8545581996E-01	0.6011378765E+00
66	0.7078369707E-01	0.3166824877E+00	0.5017414093E+00
67	0.8891370893E-01	0.3859643042E+00	0.3850632012E+00
68	0.1122341007E+00	0.1393730044E+00	0.6837037206E+00
69	0.1487948000E+00	0.1515955031E+00	0.6527392864E+00
70	0.1860724986E+00	0.2641281486E-01	0.5289260149E+00
71	0.3097940087E+00	0.4747741669E-01	0.3588519990E+00
72	0.1144608036E+00	0.1517399997E+00	0.4133580029E+00
73	0.1507880986E+00	0.1906109005E+00	0.3410502076E+00
74	0.1706172973E+00	0.2265263945E+00	0.1259271950E+00
75	0.2289942056E+00	0.3038589060E+00	0.5896532163E-01
76	0.1143215969E+00	0.3475125134E+00	0.4705806077E+00
77	0.1365052015E+00	0.3987742960E+00	0.3133349121E+00
78	0.1365693957E+00	0.5241140723E+00	0.3406164050E+00
79	0.1702775955E+00	0.5960590243E+00	0.2761180103E+00

80	0.1696981043E+00	0.2615928054E+00	0.4811651111E+00
81	0.1853505969E+00	0.3369044960E+00	0.3201222122E+00
82	0.2397855967E+00	0.2163629979E+00	0.3759418130E+00
83	0.2480811030E+00	0.3784976900E+00	0.3057712913E+00
84	0.2192613930E+00	0.2088246047E+00	0.2205650955E+00
85	0.2986275852E+00	0.2121827006E+00	0.1468642056E+00
86	0.3383885026E+00	0.2949447036E+00	0.1601980031E+00
87	0.3737956882E+00	0.4145816863E+00	0.1163380966E+00
88	0.1759153008E+00	0.4035001099E+00	0.4613820016E+00
89	0.2228980958E+00	0.4619579017E+00	0.3872157931E+00
90	0.2172852010E+00	0.6011435986E+00	0.2250604033E+00
91	0.2496989071E+00	0.6760768294E+00	0.2467515022E+00
92	0.2739233971E+00	0.4865092933E+00	0.3028185964E+00
93	0.3398965895E+00	0.5414069891E+00	0.2644341886E+00
94	0.3390182853E+00	0.5252041817E+00	0.3680981100E+00
95	0.4163025916E+00	0.6538612843E+00	0.2602193952E+00
96	0.1315259039E+00	0.4091641009E+00	0.1912034005E+00
97	0.1607993990E+00	0.5165988207E+00	0.1835025996E+00
98	0.1879366040E+00	0.4849447012E+00	0.8831258863E-01
99	0.2067928016E+00	0.6063880920E+00	0.1208055019E+00
100	0.1528825015E+00	0.6418548822E+00	0.1632298976E+00
101	0.1782497019E+00	0.7136489749E+00	0.1472460032E+00
102	0.1950799972E+00	0.7978122234E+00	0.1000023037E+00
103	0.2166976035E+00	0.8509386778E+00	0.1027066037E+00
104	0.1565562040E+00	0.6794986725E+00	0.5061172321E-01
105	0.1849167049E+00	0.7903293967E+00	0.2766529471E-01
106	0.1756449044E+00	0.8734303117E+00	0.4561759904E-01
107	0.1827957928E+00	0.9436277747E+00	0.1549258269E-01
108	0.2184236050E+00	0.8266959786E+00	0.3775115311E-01
109	0.2439989001E+00	0.8668105006E+00	0.4946492240E-01
110	0.2295816988E+00	0.9242373705E+00	0.2596106380E-01
111	0.2636404037E+00	0.9466627836E+00	0.3032221086E-01
112	0.2556335926E+00	0.5659329891E+00	0.3311180696E-01
113	0.2362577021E+00	0.7178587914E+00	0.5643908307E-01
114	0.3040370047E+00	0.6967657208E+00	0.4607859254E-01
115	0.3047299087E+00	0.7958889008E+00	0.6132747978E-01
116	0.2486615032E+00	0.7239356041E+00	0.1274670959E+00
117	0.2704665959E+00	0.8301647902E+00	0.1037508994E+00
118	0.3247472048E+00	0.8424968719E+00	0.7406390458E-01
119	0.3339362144E+00	0.8971030712E+00	0.7224391401E-01
120	0.2825644910E+00	0.8350514174E+00	0.1852613129E-01
121	0.3211824000E+00	0.8924705982E+00	0.1794643141E-01

122	0.3117949069E+00	0.9256172776E+00	0.3072093800E-01
123	0.3341934979E+00	0.9692093730E+00	0.1217324380E-01
124	0.3961125910E+00	0.8843719959E+00	0.1469997689E-01
125	0.3869144917E+00	0.9570943117E+00	0.1347938739E-01
126	0.4470236897E+00	0.9570295811E+00	0.9546336718E-02
127	0.4775207937E+00	0.9821900129E+00	0.6872606464E-02
128	0.2058212012E+00	0.6196407974E-01	0.7798926234E+00
129	0.2518475056E+00	0.1279000044E+00	0.5915088058E+00
130	0.2544569075E+00	0.3105351925E+00	0.5829390883E+00
131	0.3156026006E+00	0.3249501884E+00	0.4639602900E+00
132	0.3772808015E+00	0.4402553663E-01	0.6654961705E+00
133	0.4207046032E+00	0.8673334122E-01	0.4119533002E+00
134	0.5295364857E+00	0.1508532017E+00	0.3733662069E+00
135	0.8137475252E+00	0.1406006068E+00	0.2279590964E+00
136	0.3151867092E+00	0.2598291039E+00	0.3530336022E+00
137	0.3418394029E+00	0.3817451894E+00	0.2589038014E+00
138	0.4035350084E+00	0.3584713042E+00	0.3537184894E+00
139	0.5077272058E+00	0.3992733955E+00	0.2898977101E+00
140	0.4722937942E+00	0.2965064943E+00	0.2139941007E+00
141	0.5662766099E+00	0.4028589129E+00	0.1497109979E+00
142	0.6893994808E+00	0.3969871104E+00	0.1982202977E+00
143	0.8628751040E+00	0.4853290021E+00	0.1307560951E+00
144	0.4045999944E+00	0.1813627034E+00	0.6773204803E+00
145	0.4689007103E+00	0.2662957013E+00	0.4910367131E+00
146	0.4937984943E+00	0.4234876931E+00	0.4420625865E+00
147	0.6434162855E+00	0.5166773796E+00	0.3720596135E+00
148	0.6337016821E+00	0.1006304994E+00	0.6432744861E+00
149	0.7272452116E+00	0.2273804992E+00	0.4609940946E+00
150	0.6628788114E+00	0.3772400916E+00	0.3294067085E+00
151	0.8595455885E+00	0.4418708980E+00	0.3302043974E+00
152	0.4765160978E+00	0.5293753147E+00	0.2594738901E+00
153	0.5636020899E+00	0.6010723114E+00	0.1951211989E+00
154	0.5510414243E+00	0.6588652730E+00	0.2373663932E+00
155	0.6402565241E+00	0.7532817721E+00	0.1749967039E+00
156	0.7472360134E+00	0.5478879809E+00	0.2225959003E+00
157	0.8277941942E+00	0.6562498212E+00	0.2030532956E+00
158	0.1119096041E+01	0.6475703120E+00	0.2262544930E+00
159	0.1122905016E+01	0.7757607102E+00	0.1542932987E+00
160	0.2242027968E+00	0.4361718893E+00	0.1964779049E+00
161	0.2674177885E+00	0.5595858097E+00	0.1660642028E+00
162	0.3104830980E+00	0.5457221270E+00	0.9627965093E-01
163	0.3328841031E+00	0.6755965948E+00	0.1057367995E+00

164	0.3720475137E+00	0.5095406771E+00	0.1847946048E+00
165	0.4265367091E+00	0.5779184103E+00	0.1757698953E+00
166	0.4357596040E+00	0.6473218799E+00	0.1036375985E+00
167	0.4856935143E+00	0.7093607187E+00	0.1225507036E+00
168	0.3159776032E+00	0.6743202209E+00	0.1763204038E+00
169	0.3529244959E+00	0.7572814226E+00	0.1405256987E+00
170	0.4068664014E+00	0.7727342844E+00	0.1524548978E+00
171	0.4181675911E+00	0.8476157784E+00	0.1214092970E+00
172	0.4287396073E+00	0.7954624295E+00	0.8431659639E-01
173	0.4728245139E+00	0.8548663855E+00	0.7520925999E-01
174	0.5418804288E+00	0.8417572975E+00	0.1007888988E+00
175	0.5576245785E+00	0.9113693237E+00	0.6275516003E-01
176	0.3971770108E+00	0.7196612954E+00	0.4043109342E-01
177	0.3923479021E+00	0.8275126219E+00	0.4315078631E-01
178	0.4840024114E+00	0.7903401852E+00	0.3052827716E-01
179	0.5030732155E+00	0.8705636859E+00	0.3875757754E-01
180	0.3900350034E+00	0.8867447972E+00	0.4544848204E-01
181	0.4045810103E+00	0.9283564091E+00	0.4342215508E-01
182	0.4665615857E+00	0.9232664108E+00	0.3089918755E-01
183	0.4797903001E+00	0.9536154866E+00	0.2980542555E-01
184	0.4771012068E+00	0.8924021721E+00	0.1256007701E-01
185	0.5173798800E+00	0.9331367016E+00	0.1438540779E-01
186	0.5575615764E+00	0.9366083145E+00	0.2553104609E-01
187	0.5714020729E+00	0.9634227157E+00	0.2135989256E-01
188	0.5851451159E+00	0.9320334792E+00	0.8649736643E-02
189	0.5510594845E+00	0.9707422256E+00	0.6941570900E-02
190	0.6335691810E+00	0.9735900760E+00	0.5166618619E-02
191	0.5969625115E+00	0.9864069819E+00	0.7369614672E-02
192	0.4779720008E+00	0.4981344938E+00	0.5844748393E-01
193	0.6348072290E+00	0.5891370773E+00	0.3766107932E-01
194	0.6167864203E+00	0.6059731841E+00	0.1244375035E+00
195	0.8251013160E+00	0.6318687797E+00	0.9435034543E-01
196	0.5646904111E+00	0.7259691954E+00	0.6244398281E-01
197	0.6031528711E+00	0.7809702158E+00	0.8972237259E-01
198	0.5980842113E+00	0.8440924287E+00	0.5054383352E-01
199	0.5895571709E+00	0.8967409134E+00	0.4768311605E-01
200	0.7048380971E+00	0.7237101793E+00	0.1033923998E+00
201	0.8334618211E+00	0.7786496282E+00	0.7027542591E-01
202	0.7637972236E+00	0.7895383239E+00	0.1180671006E+00
203	0.8420091271E+00	0.8581050038E+00	0.9375898540E-01
204	0.7073963284E+00	0.8697832823E+00	0.5842054635E-01
205	0.6991103888E+00	0.9158318043E+00	0.6012412533E-01

206	0.8318998814E+00	0.9036728740E+00	0.4429496452E-01
207	0.8849071264E+00	0.9387481213E+00	0.3935790807E-01
208	0.6044886708E+00	0.8107022047E+00	0.2025584877E-01
209	0.6730198264E+00	0.8731182814E+00	0.1431398839E-01
210	0.6342272758E+00	0.9009475112E+00	0.2692351677E-01
211	0.6683933735E+00	0.9305152893E+00	0.3192029893E-01
212	0.6766291261E+00	0.9417983294E+00	0.1271538995E-01
213	0.6850581169E+00	0.9629445076E+00	0.2146667801E-01
214	0.7299829721E+00	0.9634938836E+00	0.6568749901E-02
215	0.7426728010E+00	0.9863529205E+00	0.5142178852E-02
216	0.7833564281E+00	0.8852707744E+00	0.2286950313E-01
217	0.8246064782E+00	0.9266918898E+00	0.1838962547E-01
218	0.7951936722E+00	0.9550110102E+00	0.1876292750E-01
219	0.8614364862E+00	0.9694117904E+00	0.1849077269E-01
220	0.8687235117E+00	0.9521716833E+00	0.6444055587E-02
221	0.9092599750E+00	0.9828807712E+00	0.6192821544E-02
222	0.1090209961E+01	0.9633936286E+00	0.8265390992E-02
223	0.1166924953E+01	0.9859452248E+00	0.6158319302E-02
224	0.1000375986E+01	0.1854172051E+00	0.5423095226E+00
225	0.1570129037E+01	0.2168993950E+00	0.4258900881E+00
226	0.1075256944E+01	0.3578406870E+00	0.2021154016E+00
227	0.1512421966E+01	0.4072645903E+00	0.1910302043E+00
228	0.1027004004E+01	0.4538662136E+00	0.3981646895E+00
229	0.1506876945E+01	0.5409070253E+00	0.3729160130E+00
230	0.1685479045E+01	0.5329133272E+00	0.2578333914E+00
231	0.2448146105E+01	0.6229693294E+00	0.1946762949E+00
232	0.1146139979E+01	0.6132298708E+00	0.1224573031E+00
233	0.1672626972E+01	0.6349121928E+00	0.8354412764E-01
234	0.1636142015E+01	0.7288373113E+00	0.1244362965E+00
235	0.1811290979E+01	0.8061832190E+00	0.1446519941E+00
236	0.2685538054E+01	0.3550836146E+00	0.3155787885E+00
237	0.1071245956E+02	0.4264922142E+00	0.2618879080E+00
238	0.2574728966E+01	0.8001124859E+00	0.7955052704E-01
239	0.4390925884E+01	0.8128105998E+00	0.1211687997E+00
240	0.8725733757E+00	0.7602546811E+00	0.2738693357E-01
241	0.1000702977E+01	0.8445605040E+00	0.2932630666E-01
242	0.1025166988E+01	0.8316953182E+00	0.7419369370E-01
243	0.1113782048E+01	0.8983482718E+00	0.6792701036E-01
244	0.1019778013E+01	0.9057493806E+00	0.1808738336E-01
245	0.9740859270E+00	0.9485203028E+00	0.1788486727E-01
246	0.1141852975E+01	0.9295607805E+00	0.3048870526E-01
247	0.1114433050E+01	0.9626024961E+00	0.2363116853E-01

248	0.1463426948E+01	0.7795253992E+00	0.4330257326E-01
249	0.1333399057E+01	0.8801984191E+00	0.3900365904E-01
250	0.1495571017E+01	0.9150947928E+00	0.3560945764E-01
251	0.1602329969E+01	0.9379981756E+00	0.4199227318E-01
252	0.1346436977E+01	0.9428874254E+00	0.1151810307E-01
253	0.1516757011E+01	0.9776836038E+00	0.8404225111E-02
254	0.1893331051E+01	0.9280338287E+00	0.2603170462E-01
255	0.2636862040E+01	0.9607506990E+00	0.1253264863E-01

Note that in conducting the code search Rpp correlation may be ignored in (Table 2.1.3.3.2.6.5.2–1), since it is a constant, thus eliminating the need to compute it. β_q, the quantized long term predictor coefficient, and γ_{lq}, the quantized gain for the code vector selected from the l-th codebook (l = 1,2), are reconstructed from:

$$\beta_q = \sqrt{\frac{RS\ GS_{vq}\ P0_{vq}}{R_x(0)}} \qquad (2.1.3.3.2.6.5.2\text{–}11)$$

$$\gamma_{1_q} = \sqrt{\frac{RS\ GS_{vq}\ P1_{vq}}{R_x(1)}} \qquad (2.1.3.3.2.6.5.2\text{–}12)$$

$$\gamma_{2_q} = \sqrt{\frac{RS\ GS_{vq}(1\text{-}P0_{vq}\text{-}P1_{vq})}{R_x(2)}} \qquad (2.1.3.3.2.6.5.2\text{–}13)$$

where GS_{vq}, $P0_{vq}$, and $P1_{vq}$ are the elements of the vector chosen from the {GS, P0, P1} codebook. The index of the corresponding codebook entry selected is then assigned to GSP0_x, where x is the current subframe number (1 to 4).

A special case occurs when the long term predictor is disabled for a certain subframe. This occurs when no positive correlation is found during the lag search, or when the state of the long term predictor is populated entirely by zeroes (e.g., at the first subframe).

When the long term predictor is deactivated, a modified form of (Table 2.1.3.3.2.6.5.2–1) is used:

$$E \cong R_{pp} - b\sqrt{GS\ P1} - c\sqrt{GS\ (1\text{-}P1\text{-}P0)} + f\ GS\ \sqrt{P1(1\text{-}P1\text{-}P0)} + h\ GS\ P1 + i\ GS\ (1\text{-}P1\text{-}P0)$$

(2.1.3.3.2.6.5.2–14)

For this case the quantized code vector gains are:

$$\beta_q = 0 \qquad (2.1.3.3.2.6.5.2\text{–}15)$$

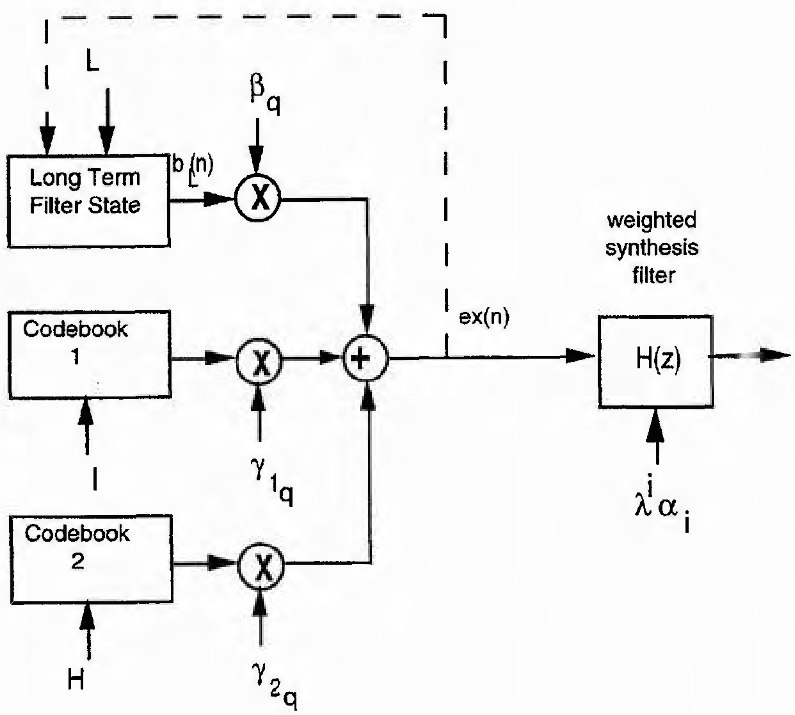
$$\gamma_{1_q} = \sqrt{\frac{RS\ GS_{vq}\ P1_{vq}}{R_x(1)}} \qquad (2.1.3.3.2.6.5.2\text{–}16)$$

$$\gamma_{2_q} = \sqrt{\frac{RS\ GS_{vq}\ (1\text{-}P1_{vq}\text{-}P0_{vq})}{R_x(2)}} \qquad (2.1.3.3.2.6.5.2\text{–}17)$$

2.1.3.3.2.6.6 Update Filter States

After all subframe parameters have been determined and quantized, the long term filter state and the weighted synthesis filter state must be updated in preparation for processing the next subframe. Figure 2.1.3.3.2.6.6–1 shows the weighted synthesizer employed in the speech encoder.

Figure 2.1.3.3.2.6.6–1 Weighted Synthesizer



The combined excitation, ex(n), shall be computed as:

$$ex(n) = \beta_q \, b_L(n) + \gamma 1_q \, u_{1,I}(n) + \gamma 2_q \, u_{2,H}(n) \qquad (2.1.3.3.2.6.6\text{–}1)$$

for $0 \le n \le 39$

The long term predictor state, r(n), is updated by:

$$r(n) = r(n+40) \quad \text{for } -146 \le n \le -41 \qquad (2.1.3.3.2.6.6\text{–}1a)$$

$$r(n) = ex(n+40) \quad \text{for } -40 \le n \le -1 \qquad (2.1.3.3.2.6.6\text{–}1b)$$

The weighted synthesis filter is updated by inputting the 40 samples of ex(n) into the weighted synthesis filter. The state of the weighted synthesis filter should reflect the filter's state at the end of the previous subframe processing prior to filtering the excitation, ex(n), for the current subframe. A direct form filter should be used for the weighted synthesis filter.

2.1.3.3.3 Channel Coding

The channel error control for alternate codecs is defined in IS-641.

The channel error control for the speech codec data defined in this standard (see Sections 2.1.3.3.3.1 to 2.1.3.3.3.4) employs three mechanisms for the mitigation of channel errors. The first is to use a rate one-half convolutional code to protect the more vulnerable bits of the speech codec data stream. The second technique interleaves the transmitted data for each speech codec frame over two time slots to mitigate the effects of Rayleigh fading. The third technique employs the use of a cyclic redundancy check over some of the most perceptually significant bits of the speech codec output. After the error correction is applied at the receiver, these cyclic redundancy bits are checked to see if the most perceptually significant bits were received properly.

2.1.3.3.3.1 Definition of Terms, Nomenclature, and Assumptions

a(X)	–	the eleventh order input polynomial to the CRC
a'(X)	–	the eleventh order CRC input polynomial at the receiver which may include the effects of channel errors
b(X)	–	the sixth order CRC parity polynomial
b'(X)	–	the sixth order CRC parity polynomial received which may include the effects of channel errors
bit position	–	in speech codec parameters the lsb is bit 0, the msb is bit n–1 where there are n bits in the parameter. i.e., the speech parameter R0 has 5 bits, the msb is bit 4, the lsb is bit 0
bit channel position	–	bits are transmitted from low to high. The first bit transmitted is bit 0, the last transmitted bit of the frame is bit 259
bit position class 1	–	bit 0 , cl1[0] is the first bit to be encoded, bit 88 , cl1[88] the last
$cc_0[i]$	–	the output of $g_0(D)$ to input bit i, cl1[i]
$cc_1[i]$	–	the output of $g_1(D)$ to input bit i, cl1[i]

CL1[i]	–	Input bit array to the convolutional encoder where i = 0..88
CL2[i]	–	class 2 bits (unencoded bits) where i ranges from 0 to 81
class 1	–	those bits which are convolutionally encoded
class 2	–	those bits which are not convolutionally encoded
crc[a'(X)]	–	the sixth order CRC parity polynomial generated from the received input bits (a'(X))
CRC	–	Cyclic Redundancy Checking code
CRC generator	–	the CRC generator polynomial
CRC's	–	the CRC parity bits, b(X)
frame X	–	the first of the two speech codec frames
frame Y	–	the speech codec frame occurring after frame X
$g_0(D)$	–	The first of the two convolutional code generator polynomials (65 octal) $g_0(D) = 1 + D + D^3 + D^5$
$g_1(D)$	–	The second of the two convolutional code generator polynomials 57 octal, $g_1(D) = 1 + D^2 + D^3 + D^4 + D^5$
$g_{crc}(X)$	–	The CRC generator polynomial $g_{crc}(X) = 1 + X + X^2 + X^4 + X^5 + X^7$
interleaving	–	Ordering of the bits on the channel.
memory order, m	–	memory order of the convolutional code, where 2^m = the number of convolutional states. For this system, m = 5
parameter names	–	the subframe information can be deciphered as follows: code2_3 is the second code vector for subframe 3
q(X)	–	the CRC quotient
subframe	–	one of the four subdivisions of a speech frame, each subframe is 5 milliseconds in duration. For each subframe, the speech codec generates 4 parameters: CODE1_X, CODE2_X, LAG_X and GSP0_X where X refers to the subframe number Refer to 2.1.3.3.2.1)

2.1.3.3.3.2 Speech Data Classes

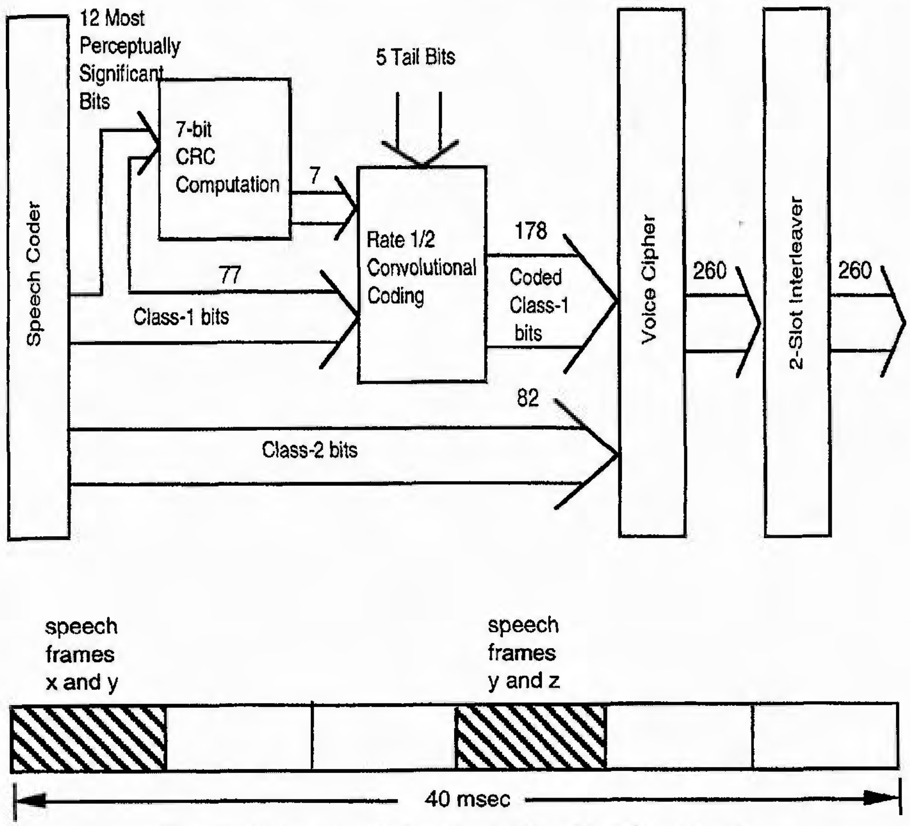
The first step in the error correction process is the separation of the 159-bit speech codec frame's information into class 1 and class 2 bits. There are 77 class 1 bits and 82 class 2 bits in the 159-bit speech codec frame. The class 1 bits represent that portion of the speech data stream to which the convolutional coding is applied. A 7-bit CRC is used for error detection purposes and is computed over the 12 most perceptually significant bits of the class 1 bits for each frame. Class 2 bits are transmitted without any error protection. The process is depicted in Figure 2.1.3.3.3.2–1. Table 2.1.3.3.3.2–1 describes the bit allocation among the classes for the parameter bits of the speech codec.

Table 2.1.3.3.3.2–1 Speech Codec Parameter Class Bit Assignments

(Note: The number in the Class 1 and most perceptually significant columns refer to the number of most significant bits of a codeword while the Class 2 column refers to the number of least significant bits of a codeword.) See Figure 2.1.3.3.3.2–1.

Parameter	Total Codeword Bits	Class 1 Bits	Class 2 Bits	Most Perceptually Significant Bits
R0	5	4	1	3
LPC1	6	4	2	3
LPC2	5	3	2	2
LPC3	5	3	2	2
LPC4	4	2	2	1
LPC5	4	1	3	1
LPC6	3	0	3	0
LPC7	3	0	3	0
LPC8	3	0	3	0
LPC9	3	0	3	0
LPC10	2	0	2	0
LAG_1	7	7	0	0
CODE1_1	7	0	7	0
CODE2_1	7	0	7	0
GSP0_1	8	8	0	0
LAG_2	7	7	0	0
CODE1_2	7	0	7	0
CODE2_2	7	0	7	0
GSP0_2	8	8	0	0
LAG_3	7	7	0	0
CODE1_3	7	0	7	0
CODE2_3	7	0	7	0
GSP0_3	8	8	0	0
LAG_4	7	7	0	0
CODE1_4	7	0	7	0
CODE2_4	7	0	7	0
GSP0_4	8	8	0	0

Figure 2.1.3.3.3.2–1 Error Correction For Speech Codec



Class 1 bits are convolutionally coded and interleaved with class 2 bits and transmitted over two time slots.

2.1.3.3.3.3 Cyclic Redundancy Check (CRC)

A 7-bit CRC is computed for the 12 most perceptually significant bits in the frame.

The generator polynomial for the CRC is:

$$g_{crc}(X) = 1 + X + X^2 + X^4 + X^5 + X^7 \qquad (2.1.3.3.3.3\text{–}1)$$

The twelve most perceptually significant bits of the frame are identified in Table 2.1.3.3.3.2–1 and they form the input polynomial. This input polynomial is defined as

$$a(X) = CL1[80]X^{11} + CL1[4]X^{10} + CL1[79]X^9 + CL1[5]X^8 + CL1[78]X^7 + CL1[6]X^6 + CL1[77]X^5 + CL1[7]X^4 + CL1[76]X^3 + CL1[8]X^2 + CL1[75]X^1 + CL1[9]X^0 \qquad (2.1.3.3.3.3\text{-}2)$$

The parity polynomial b(X) is the remainder of a the division of the input polynomial and the generator polynomial, i.e.:

$$\frac{a(X) \bullet X^7}{g_{crc}(X)} = q(X) + \frac{b(X)}{g_{crc}(X)}$$

(2.1.3.3.3.3–3)

Where q(X) is the quotient of the division, b(X) the remainder. The quotient here is discarded and only the parity bits identified in the polynomial b(X) are encoded for transmission. To facilitate the convolutional encoding of these parity bits, they are placed into the array of Class 1 bits, CL[i]. The placement of the parity bits into CL[i] is determined by the following identification for b(X):

$$b(X) = CL1[0]\,X^6 + CL1[83]\,X^5 + CL1[1]\,X^4 + CL1[82]\,X^3 + CL1[2]\,X^2 + CL1[81]\,X^1 + CL1[3]\,X^0$$

(2.1.3.3.3.3-4)

2.1.3.3.3.4 Convolutional Encoding

There are 89 bits which are input to the convolutional coder. Of these 89 bits, 77 are class 1 bits from the speech codec which are placed into CL1[4] through CL1[80]. Bits CL1[0] through CL1[3] and CL1[81] through CL1[83] are reserved for the CRC for the frame and bits CL1[84] through CL1[88] are filled with zeros corresponding to the 5 tail bits.

The bits are rearranged in the array CL1[i] as shown in Table 2.1.3.3.3.4–2. The first column indicates the location in the array where a particular bit for a parameter is to be placed starting with CL1[0]. The second column indicates the parameter and the last column indicates the bit of the parameter where bit 0 is the least significant bit.

The convolutional encoding used is a rate 1/2, memory order 5 code (R = 1/2, m = 5). There are 32 states in this code, five memory elements. Table 2.1.3.3.3.4–1 shows all the states and their outputs to a given input. The notation for the generator polynomials, $g_0(D)$ and $g_1(D)$, follow that defined by Shu Lin and Daniel Costello in **Error Control Coding: Fundamentals and Applications**, Prentice-Hall, April 1983 on page 330.

The polynomials are defined as:

$$g_0(D) = 1 + D + D^3 + D^5 \qquad (2.1.3.3.3.4\text{–}1)$$

$$g_1(D) = 1 + D^2 + D^3 + D^4 + D^5 \qquad (2.1.3.3.3.4\text{–}2)$$

The output from the convolutional coder alternates between these two polynomials starting with $g_0(D)$ being the first in each time slot. The free coefficient in the above equations is the MSB.

Table 2.1.3.3.3.4–1 Input - Output Relationship of Convolutional Coder

	INPUT = 0		INPUT = 1			INPUT = 0		INPUT = 1	
state	**g0**	**g1**	**g0**	**g1**	**state**	**g0**	**g1**	**g0**	**g1**
0	0	0	1	1	16	1	1	0	0
1	1	0	0	1	17	0	1	1	0
2	0	1	1	0	18	1	0	0	1
3	1	1	0	0	19	0	0	1	1
4	1	1	0	0	20	0	0	1	1
5	0	1	1	0	21	1	0	0	1
6	1	0	0	1	22	0	1	1	0
7	0	0	1	1	23	1	1	0	0
8	0	1	1	0	24	1	0	0	1
9	1	1	0	0	25	0	0	1	1
10	0	0	1	1	26	1	1	0	0
11	1	0	0	1	27	0	1	1	0
12	1	0	0	1	28	0	1	1	0
13	0	0	1	1	29	1	1	0	0
14	1	1	0	0	30	0	0	1	1
15	0	1	1	0	31	1	0	0	1

The convolutional encoding process may be viewed in the following manner. Initially the encoder's memory elements are cleared, i.e., the encoder starts at state 0, the bits in the class 1 buffer are read in starting at CL1[0] and concluding with bit CL1[88]. Sequentially the output from $g_0(D)$ and $g_1(D)$ are referred to as $cc_0[i]$ and $cc_1[i]$, respectively. For each input bit, CL1[i], the two output bits, $cc_0[i]$ and $cc_1[i]$, are produced. The order, i, the bits are placed into CL1[i] is indicated in Table 2.1.3.3.3.4–2.

Table 2.1.3.3.3.4–2 Bit Ordering Into Convolutional Coder

Order, i	Parameter	Bit Number	Order, i	Parameter	Bit Number
0	CRC	6	39	GSP0_2	1
1	CRC	4	40	GSP0_4	1
2	CRC	2	41	GSP0_2	0
3	CRC	0	42	GSP0_3	0
4	R0	3	43	GSP0_1	0
5	R0	2	44	GSP0_3	1
6	LPC3	4	45	GSP0_1	1
7	LPC4	3	46	GSP0_3	2
8	LPC1	3	47	GSP0_1	2
9	LPC5	3	48	GSP0_3	3
10	LAG_2	6	49	GSP0_1	3
11	LAG_4	6	50	GSP0_3	4
12	LAG_2	5	51	GSP0_1	4
13	LAG_4	5	52	LPC2	2
14	LAG_2	4	53	GSP0_4	5
15	LAG_4	4	54	GSP0_2	5
16	LAG_2	3	55	LPC4	2
17	LAG_4	3	56	GSP0_4	6
18	GSP0_2	7	57	GSP0_2	6
19	GSP0_4	7	58	GSP0_4	0
20	LAG_2	2	59	LAG_3	0
21	LAG_4	2	60	LAG_1	0
22	LAG_2	1	61	LAG_3	1
23	LAG_4	1	62	LAG_1	1
24	LAG_2	0	63	LAG_3	2
25	LAG_4	0	64	LAG_1	2
26	GSP0_1	6	65	GSP0_3	7
27	GSP0_3	6	66	GSP0_1	7
28	R0	1	67	LAG_3	3
29	GSP0_1	5	68	LAG_1	3
30	GSP0_3	5	69	LAG_3	4
31	LPC1	2	70	LAG_1	4
32	LPC3	2	71	LAG_3	5
33	GSP0_2	4	72	LAG_1	5
34	GSP0_4	4	73	LAG_3	6
35	GSP0_2	3	74	LAG_1	6
36	GSP0_4	3	75	LPC3	3
37	GSP0_2	2	76	LPC2	3
38	GSP0_4	2	77	LPC1	4

Order, i	Parameter	Bit Number	Order, i	Parameter	Bit Number
78	LPC2	4	84	Tail	0
79	LPC1	5	85	Tail	1
80	R0	4	86	Tail	2
81	CRC	1	87	Tail	3
82	CRC	3	88	Tail	4
83	CRC	5			

Note: All the tail bits, numbered 0 through 4, are equal to 0.

2.1.3.3.4 Interleaving

Interleaving for the encoded speech data for alternate codecs is described in IS-641.

Interleaving for the VSELP encoded speech data is as follows.

Before transmission, the VSELP encoded speech data is interleaved over two time slots with the speech data from adjacent speech frames. Stated another way, each time slot contains information from two speech codec frames. The speech data is placed into a rectangular interleaving array as shown in Figure 2.1.3.3.4–1. The speech data is entered into the interleaving array column-wise. The two speech frames are referred to as x and y where x is the previous speech frame and y is the present or most recent speech frame.

Figure 2.1.3.3.4–1 Interleaving Array

0x	26x	52x	78x	104x	130x	156x	182x	208x	234x
1y	27y	53y	79y	105y	131y	157y	183y	209y	235y
2x	28x	54x	80x	106x	132x	158x	184x	210x	236x
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
12x	38x	64x	90x	116x	142x	168x	194x	220x	246x
13y	39y	65y	91y	117y	143y	169y	195y	221y	247y
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
24x	50x	76x	102x	128x	154x	180x	206x	232x	258x
25y	51y	77y	103y	129y	155y	181y	207y	233y	259y

The data (ciphered or plain text) is placed into the interleaving array in a manner that intermixes the class 2 bits from the speech codec with the convolutionally coded class 1 bits. The class 2 bits are sequentially placed into the array and occupy the following numbered locations in the interleaving array:

0, 26, 52, 78

93 through 129

130, 156, 182, 208

223 through 259

The coded class 1 bits occupy the remainder of the interleaving array and also are sequentially placed into the array. The placement of the class 1 and class 2 bits in the array is shown in Figure 2.1.3.3.4–2. Figure 2.1.3.3.4–1 indicates the frame from which each bit in Figure 2.1.3.3.4–2 is taken.

Figure 2.1.3.3.4–2 Class 1 and Class 2 Interleaving

CL2[0]	CL2[1]	CL2[2]	CL2[3]	CL2[15]	CL2[41]	CL2[42]	CL2[43]	CL2[44]	CL2[56]
cc0[0]	cc1[12]	cc0[25]	cc1[37]	CL2[16]	cc1[44]	cc0[57]	cc1[69]	cc0[82]	CL2[57]
cc1[0]	cc0[13]	cc1[25]	cc0[38]	CL2[17]	cc0[45]	cc1[57]	cc0[70]	cc1[82]	CL2[58]
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
cc1[6]	cc0[19]	cc1[31]	cc0[44]	CL2[29]	cc0[51]	cc1[63]	cc0[76]	cc1[88]	CL2[70]
cc0[7]	cc1[19]	cc0[32]	CL2[4]	CL2[30]	cc1[51]	cc0[64]	cc1[76]	CL2[45]	CL2[71]
.	.	.	.	.	.	.	.	.	.
.	.	.	.	.	.	.	.	.	.
cc1[11]	cc0[24]	cc1[36]	CL2[13]	CL2[39]	cc0[56]	cc1[68]	cc0[81]	CL2[54]	CL2[80]
cc0[12]	cc1[24]	cc0[37]	CL2[14]	CL2[40]	cc1[56]	cc0[69]	cc1[81]	CL2[55]	CL2[81]

The bits in this array are then transmitted row-wise using the following algorithm:

```
do row=0,25
    do colm=0,9
        transmit{array(row,colm)}
    end do
end do
```

The ordering of the bits into the CL2[i] array is shown in Table 2.1.3.3.4–1.

1

Table 2.1.3.3.4–1 Bit Ordering Into Class 2 Array

Order, i	Parameter	Number	Order, i	Parameter	Number
0	CODE2_4	0	41	LPC6	1
1	CODE2_4	1	42	LPC5	0
2	CODE2_4	2	43	LPC5	1
3	CODE2_4	3	44	LPC5	2
4	CODE2_4	4	45	LPC4	0
5	CODE2_4	5	46	LPC4	1
6	CODE2_4	6	47	LPC3	0
7	CODE1_4	0	48	LPC3	1
8	CODE1_4	1	49	LPC2	0
9	CODE1_4	2	50	LPC2	1
10	CODE1_4	3	51	LPC1	0
11	CODE1_4	4	52	LPC1	1
12	CODE1_4	5	53	R0	0
13	CODE1_4	6	54	CODE2_2	0
14	CODE2_3	0	55	CODE2_2	1
15	CODE2_3	1	56	CODE2_2	2
16	CODE2_3	2	57	CODE2_2	3
17	CODE2_3	3	58	CODE2_2	4
18	CODE2_3	4	59	CODE2_2	5
19	CODE2_3	5	60	CODE2_2	6
20	CODE2_3	6	61	CODE1_2	0
21	CODE1_3	0	62	CODE1_2	1
22	CODE1_3	1	63	CODE1_2	2
23	CODE1_3	2	64	CODE1_2	3
24	CODE1_3	3	65	CODE1_2	4
25	CODE1_3	4	66	CODE1_2	5
26	CODE1_3	5	67	CODE1_2	6
27	CODE1_3	6	68	CODE2_1	0
28	LPC6	2	69	CODE2_1	1
29	LPC10	0	70	CODE2_1	2
30	LPC10	1	71	CODE2_1	3
31	LPC9	0	72	CODE2_1	4
32	LPC9	1	73	CODE2_1	5
33	LPC9	2	74	CODE2_1	6
34	LPC8	0	75	CODE1_1	0
35	LPC8	1	76	CODE1_1	1
36	LPC8	2	77	CODE1_1	2
37	LPC7	0	78	CODE1_1	3
38	LPC7	1	79	CODE1_1	4
39	LPC7	2	80	CODE1_1	5
40	LPC6	0	81	CODE1_1	6

2.1.3.3.5 Time Alignment

2.1.3.3.5.1 Time Alignment Process

Time Alignment is the process of controlling the time of TDMA time slot burst transmission from the mobile station by advancing or retarding the mobile station transmit burst so that it arrives at the base station receiver in the proper time relationship to other time slot burst transmissions. An error in Time Alignment is caused by the arrival of power from two different mobile station transmitters simultaneously at the base station receiver. This in turn causes errors in both signals. This overlap will occur at the head or tail of a time slot. The mechanism for detection of overlap is left to the implementor and is not subject to standardization. Upon detecting an overlap condition, the base station must send an appropriate Physical Layer Control message containing a Time Alignment information element to the mobile station using the appropriate forward signaling channel.

The format of the Physical Layer Control message is described in Section 3.7.3.1.3.2.5. The time adjustment parameter in that message provides for advancing or retarding the time of the mobile station transmit burst in units of 1/2 Symbols. Upon receipt of a Physical Layer Control message containing a Time Alignment information element, the mobile station shall change its timing in one adjustment.

2.1.3.3.5.2 Time Alignment at Initial Traffic Channel Assignment

At certain times it is necessary for a mobile station while operating on a digital traffic channel to transmit during its slot interval a SHORTENED BURST (see Section 2.1.3.3.5.4), so as to avoid collisions at the base station between the mobile station's burst and the burst of a neighboring slot. This collision of neighboring bursts at the base station is due to the mobile station not having the proper Time Alignment information corresponding to its distance from the base station.

When a mobile station receives an Initial Traffic Channel Designation (ITCD) message or a Digital Traffic Channel Designation (DTCD) message, it moves to the assigned traffic channel and proceeds to acquire synchronization according to the information contained in this message:

> If the mobile station has been assigned this traffic channel from an analog control channel (see Section 2.6.5.2), it shall proceed as follows:
>
> - The mobile station first acquires synchronization and then begins transmitting SHORTENED BURST (defined in Section 2.1.3.3.5.4) at the standard offset reference position. The mobile station continues transmitting SHORTENED BURST until it receives a Physical Layer Control message containing a Time Alignment information element from the base station or it is directed to stop transmission due to the timeout of its fade timer (see Section 2.6.5.1). If the mobile station receives a Physical Layer Control message containing a Time Alignment information element, it adjusts its transmission timing accordingly and then begins transmitting time aligned bursts (i.e., full slot duration) to the base station.

If the mobile station has been assigned this traffic channel from a digital control channel (see Section 6.4.3.6 of IS-136.1), it shall proceed as follows:

- If the DTCD indicates that SHORTENED BURST is disabled, the mobile station shall first acquire synchronization and then adjust its transmit time to reflect the standard offset reference plus the received Time Alignment information. The mobile station then begins transmitting time aligned bursts to the base station.
- If the DTCD indicates that SHORTENED BURST is enabled, the mobile station shall begin transmitting SHORTENED BURST at the standard offset reference position. The mobile station continues transmitting SHORTENED BURST until it receives a Physical Layer Control message containing a Time Alignment information element from the base station or it is directed to stop transmission due the timeout of its fade timer. If the mobile station receives a Physical Layer Control message containing a Time Alignment information element, it adjusts its transmission timing accordingly and then begins transmitting time aligned bursts to the base station.

2.1.3.3.5.3 Time Alignment at Handoff

Handoff orders contain estimated Time Alignment information used when handing off from one digital traffic channel to another. For smaller diameter cells, this estimated Time Alignment information will be used to adjust the mobile station transmit timing so that there will be no burst collisions at the base station. For systems with sector to sector handoff, the estimated Time Alignment information will also be used to adjust the mobile station transmit timing so that there will be no burst collisions at the base station. For larger diameter cells, however, this estimated Time Alignment information may not be accurate enough to avoid burst collisions at the base station.

In addition to Time Alignment information, handoff orders may also include Delta Time information (SBI = 0X), used when handing off from one digital traffic channel to another. If a handoff order indicates that SHORTENED BURST is disabled and the Delta Time information element is included, the mobile station shall proceed as follows:

- If the mobile station supports Delta Time information element, the mobile station shall:
    - Acquire synchronization as follows:
        - Adjust its transmit time (relative to its previous channel) to reflect Delta Time in order to establish a new transmit time T0.
        - The mobile station starts looking for synchronization assuming that the offset between T0 and its unknown receive time consists of the standard offset reference (SOR) plus Time Alignment information plus 162 symbols (1 time slot).
        - If the mobile station can find synchronization in the time window allowed (T0+162 symbols + SOR - 4 symbols to T0 + 162 symbols + SOR + 19 symbols), it shall consider its Time Alignment value to be the difference between T0 and the standard offset reference based transmit time specific to its current channel (i.e., directly related to its current forward DTC synchronization). If this Time Alignment value is less than 0 or greater than 30 half symbols, the mobile station shall set its Time Alignment to the value of Time Alignment received in the handoff order and then adjust its transmit

time accordingly. If this Time Alignment value is between 0 and 30 half symbols, the mobile station shall continue to use T0 for its transmit time.

- Otherwise, if the mobile station cannot acquire synchronization in the time window allowed, it shall acquire synchronization without using Delta Time information, set its Time Alignment to the Time Alignment received in the handoff order and then adjust its transmit time accordingly.

- The mobile station shall then begin transmitting time aligned bursts to the base station.

- Otherwise, if the mobile station does not support Delta Time, the mobile station shall acquire synchronization without using Delta Time, adjust its Time Alignment according to the Time Alignment information element included in the handoff order and then begin transmitting time aligned bursts to the base station.

Handoff orders also contain Shortened Burst Indicator (SBI) information (see Section 2.7.3.1.3.3) included whenever a mobile station is handed off to a digital traffic channel. A mobile station shall respond to SBI information as follows:

SBI = 00

A handoff to a small diameter cell: The mobile station first synchronizes to the forward traffic channel. The mobile station's first transmission is a full duration burst transmitted at the time derived from Timeslot Indicator, Delta Time and Time Alignment information received during the handoff order. All further transmissions to the current base station should also be full duration bursts. The mobile station adjusts its transmission timing according to Section 2.1.3.3.5.1 whenever it receives Time Alignment information from the base station.

SBI = 01

A handoff sector to sector (within the same cell.) The mobile station first synchronizes to the forward traffic channel . The mobile station's first transmission is a full duration burst transmitted at the time derived from Timeslot Indicator, Delta Time and Time Alignment information received during the handoff order. All further transmissions to the current base station should also be full duration bursts. The mobile station adjusts its transmission timing according to Section 2.1.3.3.5.1 whenever it receives Time Alignment information from the base station.

SBI = 10

A handoff to a large diameter cell. The mobile station first acquires synchronization and then begins transmitting SHORTENED BURST (defined in Section 2.1.3.3.5.4) at the standard offset reference position. The mobile station continues transmitting SHORTENED BURST until it receives a Physical Layer Control message containing a Time Alignment information element from the base station or it is directed to stop transmission due to the timeout of its fade timer (see Section 2.6.5.1). If the mobile station receives a Physical Layer Control message containing a Time Alignment information element, it adjusts its transmission timing accordingly and then begins transmitting time aligned bursts (i.e., full slot duration) to the base station.

SBI = 11

Reserved.

2.1.3.3.5.4 Shortened Burst Definition

The Shortened Burst format is shown in the following figure:

G1	R S D S D V S D W S D X S D Y S	G2

The Shortened Burst contains:

- G1: 3 symbol length guard time.
- R: 3 symbol length Ramp time.
- S: 14 symbol length Sync Word; The mobile station uses its assigned sync word.
- D: 6 symbol length CDVCC; The mobile station uses its assigned DVCC.
- G2: 22 Symbol length guard time. Note that the first 3 symbols of G2 consist of RAMP.

The fields V,W,X,Y contain bits as follows:

- V = 0000
- W = 00000000
- X = 000000000000
- Y = 0000000000000000

The above format allows determination by the base station of Timing Alignment after detection of any 2 or more sync words of the Shortened Burst. This is because the symbol interval between any two sync words in the above format is unique to the 2 sync words detected. Determination of the number of symbols between two detected sync words uniquely defines the location of the detected sync words within the received Shortened Burst.

2.1.3.3.6 Synchronization and Timing

The mobile station shall derive timing for the transmit symbol and TDMA frame and slot clocks from a common source which shall track the base station symbol rate as perceived at the mobile station receiver. The frequency tracking shall be maintained over all specified operating conditions.

2.1.4 Limitations on Emissions

2.1.4.1 Bandwidth Occupied

2.1.4.1.1 Analog Transmitters

Modulation products outside the region ±20 kHz from the carrier shall not exceed a level of 26 dB below the unmodulated carrier. Modulation products outside the region of ±45 kHz from the carrier shall not exceed a level of 45 dB below the unmodulated carrier. Modulation products outside the region of ±90 kHz from the carrier shall not exceed a level of (a) 60 dB below the unmodulated carrier, or (b) 43 plus 10 $\log_{10}$ (mean output power in Watts) dB below the unmodulated carrier, whichever is the higher level of power. Measurement techniques are defined in the current EIA IS-19, "Recommended Minimum Standards for 800-MHz Cellular Subscriber Units", and IS-137, "TDMA Cellular/PCS - Radio Interface - Minimum Performance Standards for Mobile Stations, Revision A".

2.1.4.1.2 Digital Transmitters

The emission power in either adjacent channel, centered ± 30 kHz from the center frequency, shall not exceed a level of 26 dB below the mean output power. The emission power in either alternate channel, centered ± 60 kHz from the center frequency, shall not exceed a level of 45 dB below the mean output power. The emission power in either 2nd alternate channel centered ± 90 kHz from the center frequency, shall not exceed a level of 45 dB below the mean output power or -13 dBm, whichever is the lower power.

2.1.4.2 Conducted Spurious Emissions

Refer to IS-137.

2.1.4.3 Radiated Spurious Emissions

Refer to IS-137.

2.2 Receiver

2.2.1 Frequency Parameters

2.2.1.1 Channel Spacing and Designation

2.2.1.1.1 800 MHz Operation

Channel spacing shall be 30 kHz and the dual-mode mobile station receive channel at 870.030 MHz (and the corresponding base station receive channel at 825.030 MHz) shall be termed channel number 1. The 20 MHz range of channels 1 through 666 as shown in Table 2.1.1.1.1-1 for System A and System B is basic. The additional 5 MHz of channels 667 through 799 and (wrap-around) 991 through 1023 for extending Systems A and B is mandatory. In either case, the station class mark (SCM, see Section 2.3.3) shall be set appropriately.

2.2.1.1.2 1900 MHz Operation

Channel spacing shall be 30 kHz and the mobile station receive channel at 1930.050 MHz (and the corresponding base station receive channel at 1850.010 MHz) shall be termed channel number 1. The 60 MHz range of channels 1 through 1999 is shown in Table 2.1.1.1.2-1 for Bands A through F.

2.2.2 Demodulation Characteristics

2.2.2.1 Analog Voice Signals

The demodulator is followed by the following three voice-signal processing stages:

- Receive Audio Level Adjustment
- De-emphasis
- Expandor.

Pending the generation of a complete speech transmission plan for dual-mode cellular systems, the following requirements shall be met to ensure compatibility with the transmission plan for fixed speech networks.

Audio Level Adjustment for Alternate Codecs

See IS-641.

Audio Level Adjustment for the VSELP Codec

The receive audio sensitivity shall be adjusted such that a 1 kHz modulated carrier with a ±2.9 kHz peak frequency deviation produces the same output as results from a state of $R_0 = 21$ in the VSELP codec.

2.2.2.1.1 De-Emphasis

The de-emphasis characteristic must have a nominal –6 dB per octave response between 300 and 3000 Hz.

2.2.2.1.2 Expandor

This stage is the expandor portion of a 2:1 syllabic compandor. For every 1 dB change in input level to a 1:2 expandor, the change in output level is a nominal 2 dB. The signal expansion must follow all other demodulation signal processing (including the 6 dB/octave de-emphasis and filtering). The expandor must have a nominal attack time of 3 ms and a nominal recovery time of 13.5 ms as defined by the ITU (Reference: Recommendation G162, CCITT Plenary Assembly, Geneva, May-June 1964, Blue Book, Vol. 111, P. 52). The nominal reference input level to the expandor is that corresponding to a 1000 Hz tone from a carrier with a ±2.9 kHz peak frequency deviation.

2.2.2.2 Digital Voice and Data Signals

2.2.2.2.1 Demodulation

$\pi/4$ shifted, differentially encoded quadrature phase shift keying is amenable to a number of different demodulation techniques.

2.2.2.2.2 De-Interleaving

Alternate Codecs

See IS-641.

VSELP

The data from the channel must first be de-interleaved (complying with the interleaving specified in Section 2.1.3.3.4). Each time slot contains the interleaved information from part of two speech codec frames. The nomenclature at the decoder is slightly different from the encoder, now frame x is the present speech codec frame and frame y is the next speech codec frame. Each time slot contains data for both speech frames. The received data is placed row-wise into a 26 x 10 de-interleaving array. The frame that each bit belongs to is indicated in Figure 2.1.3.3.4–1. The location in the Class 2 and Coded Class 1 arrays (CL2[i], cc0[i] and cc1[i]) that each bit in the array belongs to is shown in Figure 2.1.3.3.4–2. The correspondence between the bits in Class 2 array, CL2[i], and bits in the parameter codes is shown in Table 2.1.3.3.4–1. Once the data from the time slot is used to fill the de-interleaving array, all the data for frame x is available and the data for that frame can be decoded.

2.2.2.2.3 Convolutional Decoding

Alternate Codecs

See IS-641.

VSELP

After de-interleaving, data for one entire speech codec frame becomes available. The de-interleaved data that was convolutionally encoded must now be decoded. Any known decoding technique for convolutional codes may be used. (Refer to Section 2.1.3.3.3.4). (May be decoded using Viterbi algorithm in conjunction with the use of soft channel information.)

2.2.2.2.3.1 Cyclic Redundancy Check (CRC)

After decoding the class 1 bits, the received CRC bits are checked to determine if an error has been detected in the 12 most perceptually significant bits in each frame. A second CRC polynomial is generated at the receiver over the most perceptually significant bits, and compared to the received CRC polynomial. The second CRC polynomial is generated using received information and the CRC generator polynomial. The CRC's generator polynomial is:

$$g_{crc}(X) = 1 + X + X^2 + X^4 + X^5 + X^7 \qquad (2.2.2.2.3.1\text{–}1)$$

The decoded twelve most perceptually significant bits of the frame form the input polynomial. This decoded input polynomial is defined as:

$$a'(X) = CL1[80]X^{11}+CL1[4]X^{10} + CL1[79]X^{9} + CL1[5]X^{8} + CL1[78]X^{7} + CL1[6]X^{6} + CL1[77]X^{5}+CL1[7]X^{4}+CL1[76]X^{3}+CL1[8]X^{2}+CL1[75]X^{1}+CL1[9]X^{0}$$

(2.2.2.2.3.1–2)

Where CL1[i] are the decoded Class 1 bits. The input polynomial is divided by the CRC generator polynomial yielding a quotient and remainder polynomial:

$$\frac{a'(X) \bullet X^{7}}{g_{crc}(X)} = q(X) + \frac{crc[a'(X)]}{g_{crc}(X)}$$

(2.2.2.2.3.1–3)

Where q(x) is the quotient of the division, crc[a'(X)] the remainder. The quotient is discarded leaving the remainder crc[a'(X)], the parity polynomial. The received CRC polynomial b'(X) is derived from the decoded class 1 array using the following formula:

$$b'(X) = CL1[0]X^{6} + CL1[83]X^{5} + CL1[1]X^{4} + CL1[82]X^{3} + CL1[2]X^{2} + CL1[81]X^{1} + CL1[3]X^{0}$$

(2.2.2.2.3.1-4)

The received CRC for the frame, b'(X), is compared with crc[a'(X)]. If the two differ then an error has been detected in the twelve most perceptually significant bits for that speech frame.

2.2.2.2.3.2 Bad Frame Masking

Alternate Codecs

See IS-641.

VSELP

Based on the CRC comparison, an error in the 12 most perceptually significant bits of the speech frame may be detected. This CRC comparison failure can occur because the data was corrupted by channel errors or because a FACCH message was transmitted in place of the speech data. In either case, use of this received data for the generation of the speech signal can cause severe degradation to the speech quality. To prevent this problem, a bad frame masking strategy could be employed. The strategy described in this section may be employed.

The bad frame masking system is based on a 6 state machine. On every decode of a speech frame, the state machine can change state. State 0 occurs most often and implies that the CRC comparison was successful. State 6 implies that there were at least 6 consecutive frames which failed the CRC check. The action taken at each of these states varies as well. At state 0 no action is taken. States 1 and 2 are simple frame repeats. States 3, 4 and 5 repeat and attenuate the speech. State 6 completely mutes the speech.

The state count, with one exception, indicates how many consecutive frames had CRC comparison failures. For example, state 5 indicates 5 consecutive frames (including the current frame) have failed the comparison. The only exception is state 6 which may be preceded by an indefinite number of corrupted frames. In any state (except state 6) agreement between the received and regenerated CRC's returns the state machine to the starting state, state 0. State 6 requires two contiguous correct decodes to return to state 0. This is used to provide additional protection during prolonged intervals of very poor channel conditions which might cause the CRC to occasionally falsely indicate valid speech data.

State 0 - error free state is the normal state of the system. The state machine stays at this state unless a CRC error is detected where CRC error is defined as a disagreement of the regenerated CRC, crc[a'(X)], and the received CRC, b'(X). On each successive speech frame detected in error, the state machine moves to the next higher numbered state. As soon as the CRC detects no error for a speech frame, the machine returns to state 0 unless it was in state 6. If the machine is in state 6, two successive frames with no detected errors will cause the state machine to return to state 0 otherwise the state machine stays in state 6. In each state the following actions are followed:

State 0 – No CRC error is detected. The received decoded speech data is used.

State 1 – A CRC error has been detected in the frame. The parameter values for R(0) and the LPC bits are replaced with the corresponding values from the last frame that was in state 0. The remaining decoded bits for the frame are passed to the speech decoder without modification.

State 2 – same action is taken as in state 1.

State 3 – As in state 1 and 2, a frame repeat is done, except that the value of R(0) is modified. A 4 dB attenuation is applied to the R(0) parameter, i.e., if R0 of the last state 0 frame is greater than 2, then R0 is decremented by 2 and repeated at this lower level.

State 4 – same as state 3. R(0) is again attenuated by 4 dB, so now the level is as much as 8 dB from the original value of the R(0).

State 5 – R(0) is attenuated an additional 4 dB.

State 6 – Again the frame is repeated, but this time R(0) is set to zero, totally muting the output speech. Alternatively, comfort noise could be inserted in place of the speech signal.

2.2.2.2.4 Speech Decoding

Alternate Codecs

See IS-641.

VSELP

The speech decoder takes the 7950 bps data from the channel decoder and generates the received speech signal. The speech decoder must operate correctly in conjunction with the speech codec described in Section 2.1.3.3.2.

2.2.2.2.4.1 Definitions and Basic Definitions and Basic Decoder Parameters

See Section 2.1.3.3.2.1.

2.2.2.2.4.2 Short Term Predictor Coefficients

The short-term filter is equivalent to the traditional LPC synthesis filter. The transfer function for the short-term filter is given by:

$$A(z) = \frac{1}{1-\sum_{i=1}^{N_p} \alpha_i z^{-i}} \qquad (2.2.2.2.4.2\text{–}1)$$

The short term predictor parameters are the α_i's of the short term or synthesis filter. These are standard LPC direct form filter coefficients.

2.2.2.2.4.2.1 Decoding of Coefficients

The short term predictor coefficients are coded as quantized reflection coefficients. These codes (LPC1 – LPC10) can be decoded into the 10 reflection coefficients ($r_1 - r_{10}$) using the Table 2.1.3.3.2.4.3–1.

2.2.2.2.4.2.2 Conversion to Direct Form Coefficients

See Section 2.1.3.3.2.4.5

2.2.2.2.4.2.3 Interpolation of Coefficients

See Section 2.1.3.3.2.4.6

2.2.2.2.4.3 Frame Energy

An energy value is computed and encoded once per frame. This energy value, R(0), reflects the average signal energy in the input speech over a 20 msec. interval which is centered with respect to the middle of the fourth subframe.

Decoding of Frame Energy

The quantized value of R(0), $R_q(0)$, is determined from R0 (the transmitted code for $R_q(0)$) using equation 2.1.3.3.2.5.2–3.

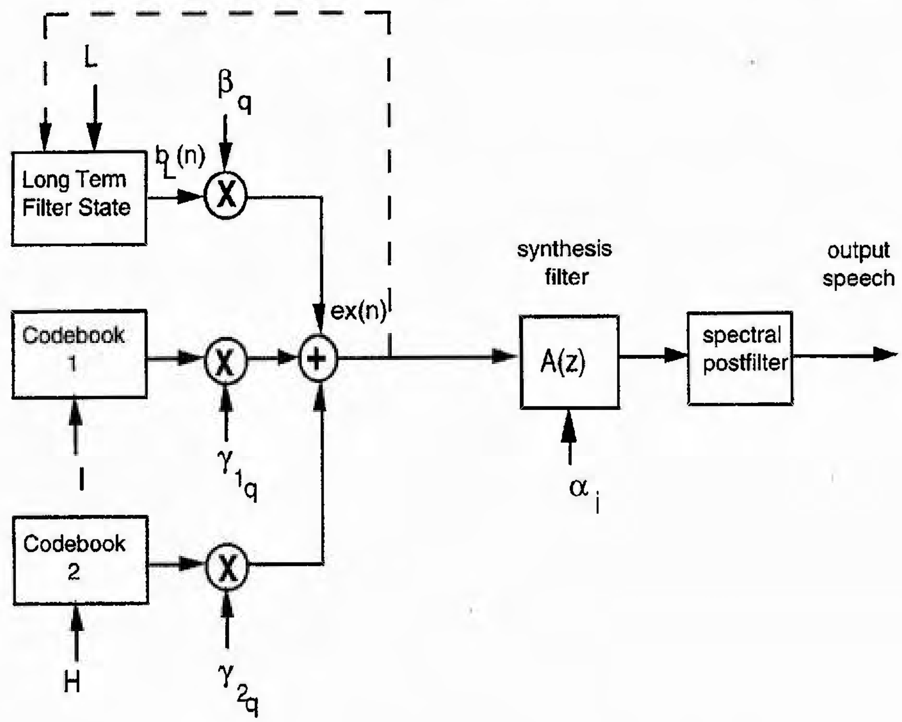
Interpolation of Frame Energy

See Section 2.1.3.3.2.5.3.

2.2.2.2.4.4 Subframe Processing

Figure 2.2.2.2.4.4–1 is a block diagram of the speech decoder.

Figure 2.2.2.2.4.4–1 Speech Decoder



2.2.2.2.4.4.1 Decoding of Lag

The lag, L, for subframe x can be determined from the code LAG_x as follows:

$$L = LAG\_x + 19 \qquad \text{if } LAG\_x \neq 0 \qquad (2.2.2.2.4.4.1\text{–}1)$$

long term predictor deactivated if LAG_x = 0.

2.2.2.2.4.4.2 Decoding of Excitation Codewords

See Section 2.1.3.3.2.6.4.4

2.2.2.2.4.4.3 Decoding of GS, P0 and P1

The GSP0_x code for subframe x is decoded in GS, P0 and P1 using Table 2.1.3.3.2.6.5.2–1.

2.2.2.2.4.4.4 Transformation of GS, P0 and P1 to Gains

GS, P0 and P1 are transformed into β_q, γ_{1q}, and γ_{2q} using equations 2.1.3.3.2.6.5.2–11, 2.1.3.3.2.6.5.2–12 and 2.1.3.3.2.6.5.2–13 if the long term predictor is activated. If the long term predictor is deactivated equations 2.1.3.3.2.6.5.2–15, 2.1.3.3.2.6.5.2–16 and 2.1.3.3.2.6.5.2–17 are used.

2.2.2.2.4.4.5 Generation of Combined Excitation

The combined excitation, ex(n), shall be computed as:

$$ex(n) = \beta_q\, b_L(n) + \gamma_{1_q}\, u_{1,I}(n) + \gamma_{2_q}\, u_{2,H}(n) \qquad \text{for } 0 \le n \le 39 \qquad (2.2.2.2.4.4.5\text{–}1)$$

where $b_L(n)$ is defined by 2.1.3.3.2.6.3.1–14 and the codebook excitation vectors, $u_{1,I}(n)$ and $u_{2,H}(n)$ are defined by 2.1.3.3.2.6.4–1 where I and H are the decoded codewords for the first and second codebooks.

2.2.2.2.4.4.6 Update Long Term Filter State

The long term predictor state, r(n), is updated by:

$$r(n) = r(n+40) \quad \text{for } -146 \le n \le -41 \qquad (2.1.3.3.2.6.6\text{–}1a)$$

$$r(n) = ex(n+40) \quad \text{for } -40 \le n \le -1 \qquad (2.1.3.3.2.6.6\text{–}1b)$$

2.2.2.2.4.4.7 Synthesis Filter

The combined excitation, ex(n), is filtered by the synthesis filter to generate the speech signal. The synthesis filter is a tenth order all pole filter. The filter coefficients for the subframe are the α_i's defined in Section 2.1.3.3.2.6.4.4. The filter coefficients will change from subframe to subframe. The filter state must be preserved from subframe to subframe. A direct form filter should be used for the synthesis filter.

2.2.2.2.4.4.8 Adaptive Spectral Postfilter

The perceptual quality of the synthetic speech may be enhanced by using an adaptive spectral postfilter as the final processing step. The form of the postfilter is:

$$\hat{H}(z) = \frac{1 - \sum_{i=1}^{10} \eta_i\, z^{-i}}{1 - \sum_{i=1}^{10} \nu^i\, \alpha_i\, z^{-i}} \;, \quad 0 \le \nu < 1 \qquad (2.2.2.2.4.4.8\text{–}1)$$

where the α_i's are the coefficients of the synthesis filter.

The numerator polynomial in 2.2.2.2.4.4.8–1 is a spectrally smoothed version of the denominator polynomial.

To derive the numerator coefficients, the autocorrelation of the impulse response of the all pole filter corresponding to the denominator of 2.2.2.2.4.4.8–1 is calculated for lags 0 through 10. The autocorrelation sequence is then windowed by a binomial window and the numerator coefficients (η_i for i = 1 to 10) are calculated from the windowed autocorrelation sequence via the Levinson recursion. Alternatively, the autocorrelation coefficients may be computed directly from the direct form coefficients via a recursion related to Levinson's recursion.

To have more control over postfiltered speech "brightness", a first order filter is used of the form:

$$\tilde{H}(z) = 1 - u\,z^{-1} \qquad (2.2.2.2.4.4.8\text{–}2)$$

This filter is cascaded with filter 2.2.2.2.4.4.8–1 and is considered part of the adaptive spectral postfilter.

The following postfilter parameters may be used. Note that B_{eq} is the bandwidth expansion factor which specifies the degree of smoothing which is performed on the denominator, to generate the numerator.

$v = 0.8$

$B_{eq} = 1200\text{Hz}$

$u = 0.4$

The spectral smoothing coefficients (autocorrelation window), wp(i), for 1200 Hz are:

wp(0)	1.000000
wp(1)	0.923077
wp(2)	0.725275
wp(3)	0.483516
wp(4)	0.271978
wp(5)	0.127990
wp(6)	0.049774
wp(7)	0.015718
wp(8)	0.003930
wp(9)	0.000748
wp(10)	0.000102

(2.2.2.2.4.4.8–3)

In order to reduce the computations needed to compute the spectrally smoothed numerator coefficients, one may perform the spectral smoothing operation once per frame on the denominator coefficients corresponding to the uninterpolated coefficients. This will yield the coefficients for the numerator of the spectral postfilter for subframe four. The numerator coefficients for subframes one, two, and three are interpolated using the same interpolation scheme that is used for the LPC synthesis coefficients as described in Section 2.1.3.3.2.4.6.

To ensure unity power gain between the input, $\hat{s}(n)$, and the output, $\hat{s}_p(n)$, of the spectral postfilter, a gain scale factor is computed and is used to scale the spectral postfiltered signal. S_{scale}, the postfilter scale factor, is:

$$S_{scale} = \sqrt{\frac{\sum_{n=0}^{N-1} \hat{s}^2(n)}{\sum_{n=0}^{N-1} \hat{s}_p^{\,2}(n)}} \qquad (2.2.2.2.4.4.8\text{–}4)$$

The scale factor, S_{scale}, is the square root of the ratio of the input signal energy to output signal energy over the subframe. This scale factor is not used directly. The scale factor is passed through a first order low pass filter. This filtering is given by:

$$S'_{scale}(n) = (.9875 * S'_{scale}(n-1)) + (.0125 * S_{scale}) \qquad (2.2.2.2.4.4.8\text{–}5)$$

The output of the spectral postfilter, $\hat{s}_p(n)$, is then multiplied by S'_{scale} as the last step in reconstructing the speech signal in the speech decoder.

2.2.2.2.4.5 Audio Interface

The function of the audio interface at the mobile station receiver is to convert the signal from the speech decoder to an analog speech signal.

The speech codec shall be succeeded by the following voice processing stages:

- Digital to Analog Converter
- Reconstruction filter
- Receive Level Adjustment

The characteristics of these stages are described in the following sections.

2.2.2.2.4.5.1 Digital to Analog Converter

The D/A converter shall be performed according to either of the following:

- by direct conversion from PCM to analog
- or by making an uniform/μlaw code conversion succeeded by a standard codec D/A converter.

The uniform/μlaw code conversion is performed according to definition in ITU (formerly CCITT) Red Book G.721, Section 4.2.7 sub-block COMPRESS. The parameter LAW shall be set to LAW = 0. The D/A conversion is based on the standard specified in ITU (formerly CCITT) Red Book G.711.

2.2.2.2.4.5.2 Reconstruction Filter

The function of the filter is to reconstruct the analog band-limited speech signal from the D/A converter. The attenuation of the filter shall comply with ITU (formerly CCITT) Red Book G.714 receiving filter.

Note: The filter specification in G.714 is concerned with PCM equipment. In some cases more attenuation will be needed in a terminal equipment.

2.2.2.2.4.5.3 Receive Level Adjustment

Pending the generation of a complete speech transmission plan for these systems, the following requirements shall be met to ensure compatibility with the transmission plan for fixed speech networks.

The receive audio sensitivity of the mobile station in the digital mode shall meet the requirements stated in IS-137 under "Digital Receiver Audio Sensitivity".

2.2.2.2.5 Delay Interval Requirements

The mobile stations and the base station shall, as a default on a digital traffic channel, provide for delay interval compensation of up to 1 symbol length. The Delay Interval Compensation (DIC) function can then be turned on or off in the mobile station by a Physical Layer Control message from the base station. The delay interval, as previously defined, is defined as the difference in µsec between the first and last ray, using a two-ray model, where both rays are of equal magnitude.

Note: The current specification is based on incomplete information on delay spread profiles found in existing cellular systems. As such, it is subject to revision and change in the future, if this is found to be necessary in light of further data.

2.2.3 Limitations on Emissions

2.2.3.1 Conducted Spurious Emissions

2.2.3.1.1 Suppression Inside Cellular/PCS Band

Any RF signals emitted in the mobile station's receive band must not exceed –80 dBm, as measured at the antenna connector. Additionally, signals in the mobile station's transmit band must not exceed –60 dBm, as measured at the antenna connector.

2.2.3.1.2 Suppression Outside Cellular/PCS Band

Refer to IS-137.

2.2.3.2 Radiated Spurious Emissions

Refer to IS-137.

2.2.4 Other Receiver Parameters

System performance is predicated upon receivers meeting IS-137, "TDMA Cellular/PCS - Radio Interface - Minimum Performance Standards for Mobile Stations, Revision A".

2.3 Security and Identification

2.3.1 Mobile Identification Number

A 34-bit binary mobile identification number (MIN) is derived from the mobile station's 10-digit directory telephone number by the following procedure.

	$MIN2_p$	$MIN1_p$		
	NPA	NXX	X	XXX
Bits	10	10	4	10

(1) The first three digits are mapped into 10 bits (corresponding to $\{MIN2_p\}$) by the following coding algorithm:

(a) Represent the 3-digit field as $D_1 D_2 D_3$ with the digit 0 having the value 10.

(b) Compute $100D_1 + 10D_2 + D_3 - 111$.

(c) Convert the result in step (b) to binary by a standard decimal-to-binary conversion (see table below).

(2) The second three digits are mapped into the 10 most significant bits of $MIN1_p$ by the coding algorithm described in (1).

(3) The last four digits are mapped into the 14 least-significant bits of $MIN1_p$ as follows:

(a) The thousands digit should be mapped into four bits by a Binary-Coded-Decimal (BCD) conversion, as specified in the table below.

(b) The last three digits are mapped into 10 bits by the coding algorithm described in (1).

Decimal-to-Binary Conversion		Thousands-Digit BCD Mapping Procedure	
Decimal Number	**Binary Number**	**Thousands Digit**	**Binary Sequence**
1	0000000001	1	0001
2	0000000010	2	0010
3	0000000011	3	0011
4	0000000100	4	0100
		5	0101
		6	0110
		7	0111
998	1111100110	8	1000
999	1111100111	9	1001
		0	1010

In the following example, the 10-digit directory telephone number 321-456-7890 is encoded into MIN2 and MIN1 using the procedure described above:

- MIN2. The 10-bit MIN2 is derived from the first three digits of the telephone number (i.e., 321):

  (a) $D_1 = 3; D_2 = 2; D_3 = 1$.

  (b) $100\,D_1 + 10\,D_2 + D_3 - 111 = 100(3) + 10(2) + (1) - 111 = 210$.

  (c) 210 in binary is 00 1101 0010.

Therefore MIN2 is 00 1101 0010.

- MIN1. The 10 most significant bits of MIN1 are derived from the second three digits of the telephone number (i.e., 456):

  (a) $D_1 = 4; D_2 = 5; D_3 = 6$

  (b) $100\,D_1 + 10\,D_2 + D_3 - 111 = 100(4) + 10(5) + (6) - 111 = 345$.

  (c) 345 in binary is 0101 0110 01.

The next four most significant bits of MIN1 are derived from the thousands digit of the telephone number (i.e., 7) by BCD conversion:

  (a) 7 in BCD is 0111.

The 10 least significant bits of MIN1 are derived from the last three digits of the telephone number (i.e., 890):

  (a) $D_1 = 8; D_2 = 9; D_3 = 10$.

  (b) $100\,D_1 + 10\,D_2 + D_3 - 111 = 100(8) + 10(9) + (10) - 111 = 789$.

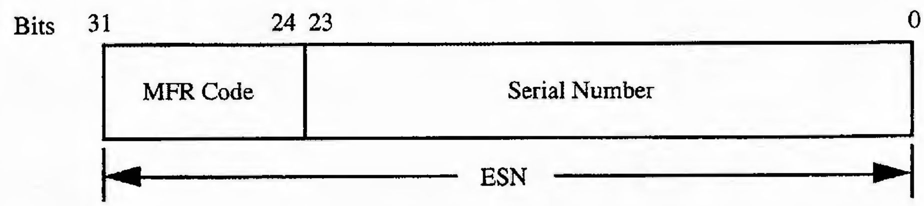
  (c) 789 in binary is 11 0001 0101.

Therefore MIN1 is 0101 0110 0101 1111 0001 0101.

2.3.2 Electronic Serial Number (ESN)

The ESN is a 32-bit binary number that uniquely identifies the mobile station to any cellular/PCS system. It must be factory-set and not readily alterable in the field. Modification of the ESN will require a special facility not normally available to subscribers. The circuitry that provides the ESN must be isolated from fraudulent contact and tampering. Electronic storage devices mounted in sockets or connected with a cable are deemed not to comply with this requirement. Attempts to change the ESN circuitry must render the mobile station inoperative.

The bit allocation of the ESN shall be as follows:



At the time of issuance of initial type acceptance, the manufacturer shall be assigned a Manufacturer's (MFR) Code within the eight most-significant bits (bit 31 through bit 24) of the 32-bit serial number. Bits 23 through 0 shall be uniquely assigned by each manufacturer.

2.3.3 Station Class Mark

Class-of-station information referred to as the station class mark (SCM_p) must be stored in a mobile station. The digital representation of this class mark is specified in the table below. A mobile station operating at 800 MHz shall support a bandwidth of 25 MHz.

Table 2.3.3-1 Station Class Marks (Bits 4-0)

Power Class (see 2.1.2.2)	**SCM_p**	**Transmission** (see 2.3.11)	**SCM_p**	**Bandwidth** (see 2.1.1.1 and 2.2.1.1)	**SCM_p**
Class I	0XX00	Continuous	XX0XX	20 MHz	X0XXX
Class II	0XX01	Discontinuous	XX1XX	25 MHz	X1XXX
Class III	0XX10				
Class IV	0XX11				
Class V	1XX00				
Class VI	1XX01				
Class VII	1XX10				
Class VIII	1XX11				

Note: In order to maintain compatibility between dual-mode mobile stations and EIA/TIA-553 base stations, dual-mode mobile stations with power classes IV - VIII must set the power class bits of the SCM to look like a Power Class III mobile station (i.e., SCM(4-0) = 0XX10) when PCI = 0 in the System Parameter Overhead message (3.7.1.2.1).

2.3.4 Registration Memory

2.3.4.1 Autonomous Registration Memory

A single 21-bit (20 data bits plus an overflow bit) Next Registration indicator ($NXTREG_{s-p}$) and corresponding 15-bit System Identification indicator (SID_{s-p}) pair must be retained when the mobile station power is turned off. The data retention time under power-off condition must be longer than 48 hours. If the integrity of the stored data can not be guaranteed after the mobile station is disconnected from the vehicle battery, then the memory must be set to zero when power is re-applied to the mobile station.

2.3.4.2 Location Area Memory

A 12-bit Location Area identifier ($LOCAID_{s-p}$) must be stored in the mobile station and used to identify changes in location area (see Section 2.6.2.1). The $LOCAID_{s-p}$ value must be retained when the mobile station power is turned off. The data retention time under power-off condition must be longer than 48 hours. If the integrity of the stored data cannot be guaranteed after the mobile station is disconnected from the vehicle battery, then the memory must be set to zero when power is re-applied to the mobile station.

A 1-bit Power-up Registration identifier ($PUREG_{s-p}$) must be stored in the mobile station and used to identify changes in the Power-up Registration flag (see Section 2.6.2.1). The $PUREG_{s-p}$ value must be retained when the mobile station power is turned off. The data retention time under power-off condition must be longer than 48 hours. If the integrity of the stored data cannot be guaranteed after the mobile station is disconnected from the vehicle battery, then the memory must be set to zero when power is re-applied to the mobile station.

2.3.5 Access Overload Class

A 4-bit Number indicator ($ACCOLC_p$) must be stored in the mobile station and used to identify which overload class field controls access attempts by the mobile station (see Section 2.6.3.4).

2.3.6 Extended Address Method

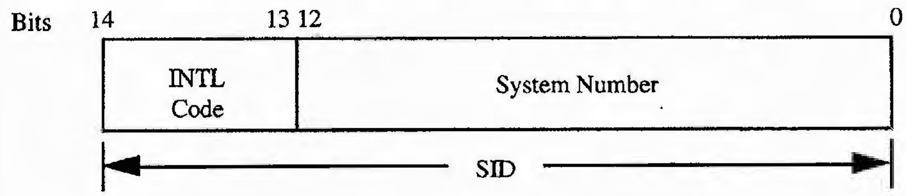
A 1-bit Access Method indicator (EX_p) must be stored in the mobile station and used to determine if the extended address word must be included in all access attempts (see Section 2.6.3.7).

2.3.7 First Paging Channel

Contains two 11-bit first paging channels ($FIRSTCHP_{p-pri}$ and $FIRSTCHP_{p-sec}$) which must be stored in the mobile station and used to identify the channel number of the first paging channel when the mobile station is "home" (see Section 2.6.1.1.2).

2.3.8 Home System Identification

A 15-bit System Identification indicator (SID_p) must be stored in the mobile station and used to identify the mobile station's home system (see Section 2.6.1.1.2). The bit allocation of the System Identification indicator (SID) shall be as follows:



The international (INTL) codes (bits 14 and 13) shall be allocated as follows:

BIT 14	BIT 13	Country
0	0	United States
0	1	Other countries
1	0	Canada
1	1	Mexico

Bits 12 through 0 will be assigned to each U.S. system by the FCC. (See Note 13 in Preface.)

2.3.9 Local Control Option

A means must be provided within the mobile station to enable or disable the local control option.

2.3.10 Preferred-System Selection (800 MHz)

A means must be provided within the mobile station to identify the preferred system as either System A or System B.

2.3.11 Discontinuous-Transmission

Discontinuous-Transmission refers to the ability of certain mobile stations to switch autonomously between two transmitter power-level states (DTX-High and DTX-Low).

2.3.11.1 Discontinuous-Transmission on an Analog Voice Channel

In the DTX-High state, the transmitter radiates at the power level indicated by the most recent power-controlling order (initial-voice-channel-designation, handoff, or power-change order) received by the mobile station. In this state the mobile station must transpond SAT at all times, except for the normal suspensions of SAT covered in Section 2.4.1.

In the DTX-Low state, the transmitter radiates at a power level determined by the DTX-High state power level (DTX-High level) and the DTX_S indicator that is copied from the DTX field in Word 2 of the System Parameter Overhead message (see Section 3.7.1.2.1). If the DTX_S indicator is set to 10, the DTX-Low level must equal or exceed a level that is 8 dB below the DTX-High level. If the DTX_S indicator is set to 11, no minimum applies to the DTX-Low level; that is, the transmitter may be turned off or it may be turned on at any level up to the DTX-High level. In the DTX-Low state, the mobile station must not transpond SAT. If the DTX_S indicator is set to 00, only the DTX-High state (that is Continuous-Transmission) is permitted. The DTX_S indicator setting of 01 is reserved.

When a mobile station switches from the DTX-High state to the DTX-Low state, it must pass through a transition state in which the transmitted power is at the DTX-High level but SAT is not transponded. The sequence must be as follows: starting in the DTX-High state, enter the transition state; remain in the transition state 300 ms; enter the DTX-Low state.

When a mobile station switches from the DTX-Low state to the DTX-High state, it must begin transponding SAT immediately after changing the power level, except for the normal suspensions of SAT covered in Section 2.4.1. Each time that the mobile station enters the DTX-High state, it must remain in that state for at least 1.5 seconds, unless it enters the DTX-High state in response to an audit order in which case it must remain in that state for at least 5 seconds. (Note that any requirement for the mobile station to remain in the DTX-High state for a certain minimum time interval does not prohibit the mobile station from leaving the conversation state before the interval ends.)

2.3.11.2 Discontinuous-Transmission on a Digital Traffic Channel

In the DTX-High state, the transmitter radiates at a power level indicated by the most recent power-controlling order (Initial Traffic Channel Designation message, Digital Traffic Channel Designation message, Handoff message, Dedicated DTC Handoff message, or Physical Layer Control message) received by the mobile station. In this state, the mobile station will send CDVCC at all times.

In the DTX-Low state, the transmitter will remain off and the CDVCC will not be sent except for the transmission of FACCH messages. All SACCH messages to be transmitted by the mobile station while in the DTX-Low state will be sent as an FACCH message after which the transmitter will return to the off state unless Discontinuous-Transmission has been otherwise inhibited.

When a mobile station switches from the DTX-High state to the DTX-Low state, it must pass through a transition state in which the transmitted power is at the DTX-High level until all pending FACCH messages in the mobile station have been entirely transmitted.

When a mobile station desires to switch from the DTX-High state to the DTX-Low state, it may complete all in-progress SACCH messages in the DTX-High state, or terminate SACCH message transmission and resend the interrupted SACCH messages, in their entirety, as FACCH messages in the DTX-Low state.

2.3.12 Authentication, Encryption of Signaling Information/User Data and Privacy

Messages received during the Authentication procedures that are unrelated to the authentication process shall also be processed.

2.3.12.1 Authentication

The term Authentication refers to the process during which information is exchanged between a mobile station and the base station for the purposes of enabling the base station to confirm the identity of the mobile station. In short, a successful outcome of the authentication process occurs only when it can be demonstrated that the mobile station and base station possess identical sets of Shared Secret Data (SSD).

2.3.12.1.1 Shared Secret Data (SSD)

SSD is a 128-bit pattern stored in the mobile station (in semi-permanent memory) and readily available to the base station. As depicted in Figure 2.3.12.1.1–1, SSD is partitioned into two distinct subsets. Each subset is used to support a different process.

Figure 2.3.12.1.1-1 Partitioning of SSD

Contents	SSD-A	SSD-B
Length (bits)	64	64

Specifically,

SSD-A is used to support the Authentication procedures; and

SSD-B is used to support voice privacy and message confidentiality.

SSD is generated according to the procedure specified in Section 2.3.12.1.9.

2.3.12.1.2 Random Challenge Memory (RAND)

A 32-bit value held in the mobile station. $RAND_S$ is used in conjunction with SSD-A and other parameters, as appropriate, to authenticate mobile station originations, terminations and registrations.

In the Forward Analog Control Channel, RAND is the concatenation of the last RAND1 A and RAND1 B values received in Random Challenge A and Random Challenge B Global Action messages appended to the overhead message train. Both RAND1 A and RAND1 B must be received on the same control channel and in the same Overhead Message Train in order for a valid RAND to exist (see Section 3.7.1.2.2).

In the Forward Digital Control Channel, RAND is received in the Access Parameters message (see IS-136.1, Section 6.4.1.1.2).

2.3.12.1.3 Call History Parameter ($COUNT_{S-p}$)

A modulo-64 count held in the mobile station. $COUNT_{S-p}$ is updated at the mobile station upon either:

- receipt of a Parameter Update Order (see Table 3.7.1–1) on the FVC
- receipt of a Parameter Update message on the SPACH (see Section 6.4.3.10 of IS-136.1); or
- receipt of a Parameter Update message on the FDTC (see Section 3.7.3.1.3.2.15).

2.3.12.1.4 MIN1 and MIN2

The 24-bit parameter referred to as MIN1 in the Authentication procedures is derived as indicated in Section 8.1.4.2 of IS-136.1. The 8-bit parameter referred to as MIN2 in the Authentication procedures is derived as indicated in Section 8.1.4.2 of IS-136.1.

2.3.12.1.5 Authentication of Mobile Station Registrations

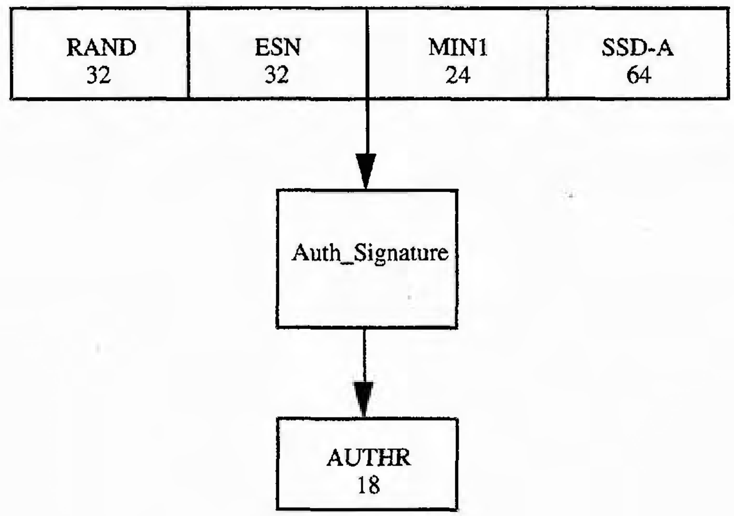
When the information element AUTH in the System Parameter Overhead message is set to 1, and the mobile station attempts to register, the following authentication-related procedures shall be performed:

- In the mobile station:
  - Initialize the authentication algorithm (Auth_Signature) as illustrated in Figure 2.3.12.1.5–1.
  - Execute the Auth_Signature procedure (see Section 2.3.12.1.11).
  - Set AUTHR equal to the 18 bits of Auth_Signature algorithm output.
  - Send AUTHR together with RANDC (eight most significant bits of RAND) and $COUNT_{S-p}$ to the base station (Authentication Word C of RECC Autonomous Registration Order message).

- At the base station:
  - Compare the received values for RANDC, and optionally COUNT, with the internally stored values associated with the received MIN1/ESN.
  - Compute AUTHR as described above, except use the internally stored value of SSD-A.
  - Compare the value for AUTHR computed internally with the value of AUTHR received from the mobile station.

If any of the comparisons by the base station fail, the base station may deem the registration attempt unsuccessful, initiate the Unique Challenge-Response procedure (see Section 2.3.12.1.6), or commence the process of updating the SSD (see Section 2.3.12.1.9).

Figure 2.3.12.1.5-1 Computation of AUTHR for Authentication of Mobile Station Registration



2.3.12.1.6 Unique Challenge-Response Procedure

The Unique Challenge-Response Procedure is initiated by the base station and can be carried out over any combination of control, traffic and/or voice channels.

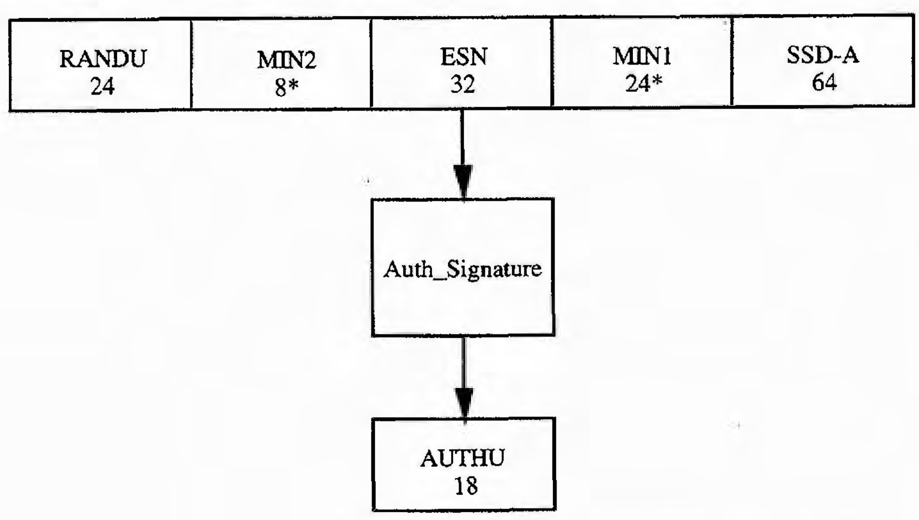
More specifically:

- At the base station:
  - A 24-bit, random pattern referred to as RANDU is generated and sent to the mobile station via:
    - the FOCC in Word 3-Unique Challenge Order Word of a mobile station control message if the procedure is to be initiated on a forward control channel (see Sections 3.6.2.3 and 3.7.1.1); or
    - the FDTC in a Unique Challenge Order FACCH message if the mobile station has been assigned to a digital traffic channel (see Sections 3.6.5 and 3.7.3.1.3.2.17); or
    - the FVC in Word 2-Unique Challenge Order Word of a mobile station control message (see Sections 3.6.4 and 3.7.2.1).
  - Initialize Auth_Signature as illustrated in Figure 2.3.12.1.6-1.
  - Execute the Auth_Signature algorithm (see Section 2.3.12.1.11).
  - Set AUTHU equal to the 18 bits of the Auth_Signature algorithm output.

- At the mobile station:
  - Compute AUTHU as described above using the received RANDU and its internally stored values for the remaining input parameters.
  - Send AUTHU to the base station via:
    - the RECC in WORD C-Unique Challenge Order Confirmation Word of an order confirmation message if the mobile station is not tuned to an analog voice or digital traffic channel (see Sections 2.6.2.3 and 2.7.1.1); or
    - the RDTC in a Unique Challenge Order Confirmation FACCH message if the mobile station is tuned to a digital traffic channel (see Sections 3.6.5 and 2.7.3.1.3.2.18); or
    - the RVC in a Unique Challenge Order Confirmation message if the mobile station is tuned to an analog voice channel.

Upon receipt of the Unique Challenge Order Confirmation from the mobile station, the base station compares the received value for AUTHU to that generated/stored internally. If the comparison fails, the base station may deny further access attempts by the mobile station, drop the call in progress, or initiate the process of updating the SSD (see Section 2.3.12.1.9).

Figure 2.3.12.1.6-1 Computation of AUTHU for Unique Challenge-Response Procedure



* See Section 8.1.4.2 of IS-136.1

2.3.12.1.7 Authentication of Mobile Station Originations

When the information element AUTH in the System Parameter Overhead message is set to 1, and the mobile station attempts to originate a call, the following authentication-related procedures shall be performed:

- In the mobile station,
    - Initialize Auth_Signature as illustrated in Figure 2.3.12.1.7-1.

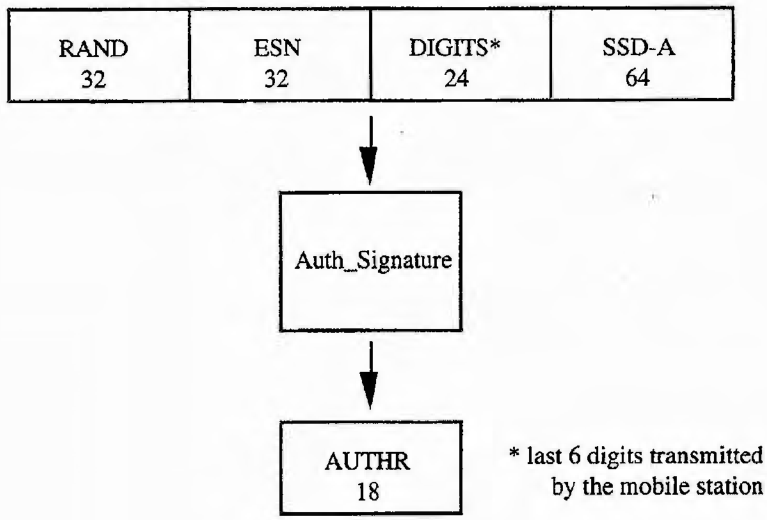
      If there were at least six digits dialed, then the last six dialed digits shall comprise the DIGITS input parameter. If there were less than six digits dialed, then the DIGITS input parameter shall be populated as follows:

        - MIN1, which is derived according to Section 8.1.4.2 of IS-136.1, shall be used to initially fill the DIGITS input parameter.
        - the least significant 4 bits of the DIGITS input parameter are replaced by the last dialed digit.
        - the next least significant 4 bits of the DIGITS input parameter are replaced by the second last dialed digit.
        - continue replacing 4-bit segments of the DIGITS input parameter in this manner until all dialed digits have been included.
    - Execute the Auth_Signature algorithm (see Section 2.3.12.1.11).
    - Set AUTHR equal to the 18 bits of the Auth_Signature algorithm output.
    - Send AUTHR together with RANDC (eight most significant bits of RAND) and $COUNT_{S-p}$ to the base station (Authentication Word C of the RECC Origination message).
- At the base station:
    - Compare the received values for RANDC, and optionally COUNT, with the internally stored values associated with the received MIN1/ESN.
    - Compute AUTHR as described above, except use the internally stored value of SSD-A.
    - Compare the value for AUTHR computed internally with the value of AUTHR received from the mobile station.

If the comparisons at the base station are successful, the appropriate channel assignment procedures are commenced. Once assigned to an analog voice or digital traffic channel, the base station may, at the discretion of the system operator, issue a Parameter Update Order (see Table 3.7.1–1) to the mobile station on the FVC or a Parameter Update message (see Section 3.7.3.1.3.2.15) on the FDTC. Mobile stations confirm the receipt of Parameter Update Orders by sending Parameter Update Confirmations on the RVC, and acknowledge receipt of Parameter Update messages via Parameter Update ACK messages sent on the RDTC.

If any of the comparisons by the base station fail, the base station may deny service, initiate the Unique Challenge-Response procedure (see Section 2.3.12.1.6), or commence the process of updating the SSD (see Section 2.3.12.1.9).

Figure 2.3.12.1.7-1 Computation of AUTHR for Authentication of Mobile Station Originations



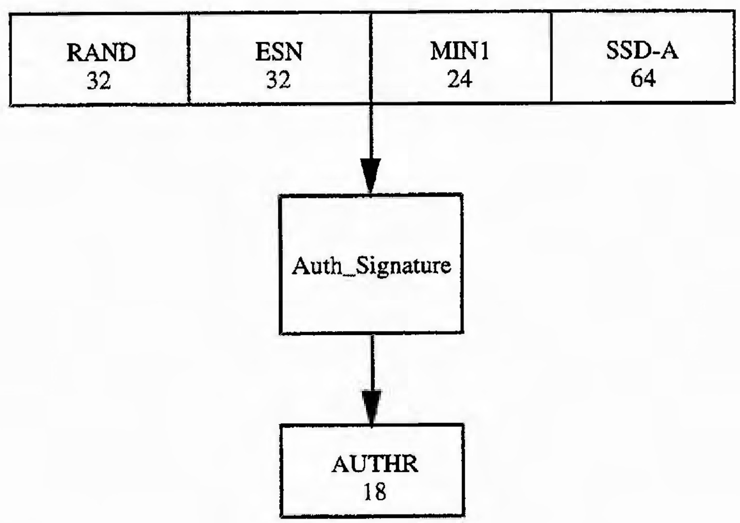
2.3.12.1.8 Authentication of Mobile Station Terminations

When the information element AUTH in the System Parameter Overhead message is set to 1, and a Page Match occurs, the following authentication-related procedures shall be performed:

- In the mobile station:
  - Initialize Auth_Signature as illustrated in Figure 2.3.12.1.8–1.
  - Execute the Auth_Signature algorithm (see Section 2.3.12.1.11).
  - Set AUTHR equal to the 18 bits of the Auth_Signature algorithm output.
  - Send AUTHR together with RANDC (eight most significant bits of RAND) and $COUNT_{S-P}$ to the base station (Authentication Word C of the RECC Page Response message).
- At the base station:
  - Compare the received values for RANDC, and optionally COUNT, with the internally stored values associated with the received MIN1/ESN.
  - Compute AUTHR as described above, except use the internally stored value of SSD-A.
  - Compare the value for AUTHR computed internally with the value of AUTHR received from the mobile station.

If the comparisons at the base station are successful, the appropriate channel assignment procedures are commenced. Once assigned to an analog voice or digital traffic channel, the base station may, at the discretion of the system operator, issue a Parameter Update Order (see Table 3.7.1–1) to the mobile station on the FVC or a Parameter Update message (see Section 3.7.3.1.3.2.15) on the FDTC. Mobile stations confirm the receipt of Parameter Update Orders by sending Parameter Update Confirmations on the RVC, and acknowledge receipt of Parameter Update messages via Parameter Update ACK messages sent on the RDTC.

Figure 2.3.12.1.8-1 Computation of AUTHR for Authentication of Mobile Station Terminations



If any of the comparisons by the base station fail, the base station may deny service, initiate the Unique Challenge-Response procedure (see Section 2.3.12.1.6), or commence the process of updating the SSD (see Section 2.3.12.1.9).

2.3.12.1.9 Updating the Shared Secret Data (SSD)

Updating the SSD involves the application of SSD_Update_Procedure (see Section 2.3.12.1.11), initialized with mobile station specific information, random data and the mobile station's A-key.

The A-key is:

- 64 bits long;
- assigned to the mobile station;
- stored in the mobile station's permanent security and identification memory; and
- is known only to the mobile station and its associated HLR/AC.

Notes:

1. The last item in the above list is intended to enhance the security of the mobile station's secret data by eliminating the need to pass the A-key itself from system to system as the subscriber roams. As a consequence, SSD updates are carried out only in the mobile station and its associated HLR/AC, not in the Serving-System. For any instances in Section 2.3.12, where it is implied that base stations update SSD, in fact, this function is carried out in the HLR/AC. The Serving-System obtains a copy of the SSD computed by the HLR/AC via intersystem communication (see EIA/TIA IS-41) with the mobile station's HLR/AC.
2. Since the SSD Update procedure involves multiple transactions and can be started on one channel and completed on another channel, call processing and signaling text above and beyond that normally included in this portion of the document has been included here for the sake of added clarity.

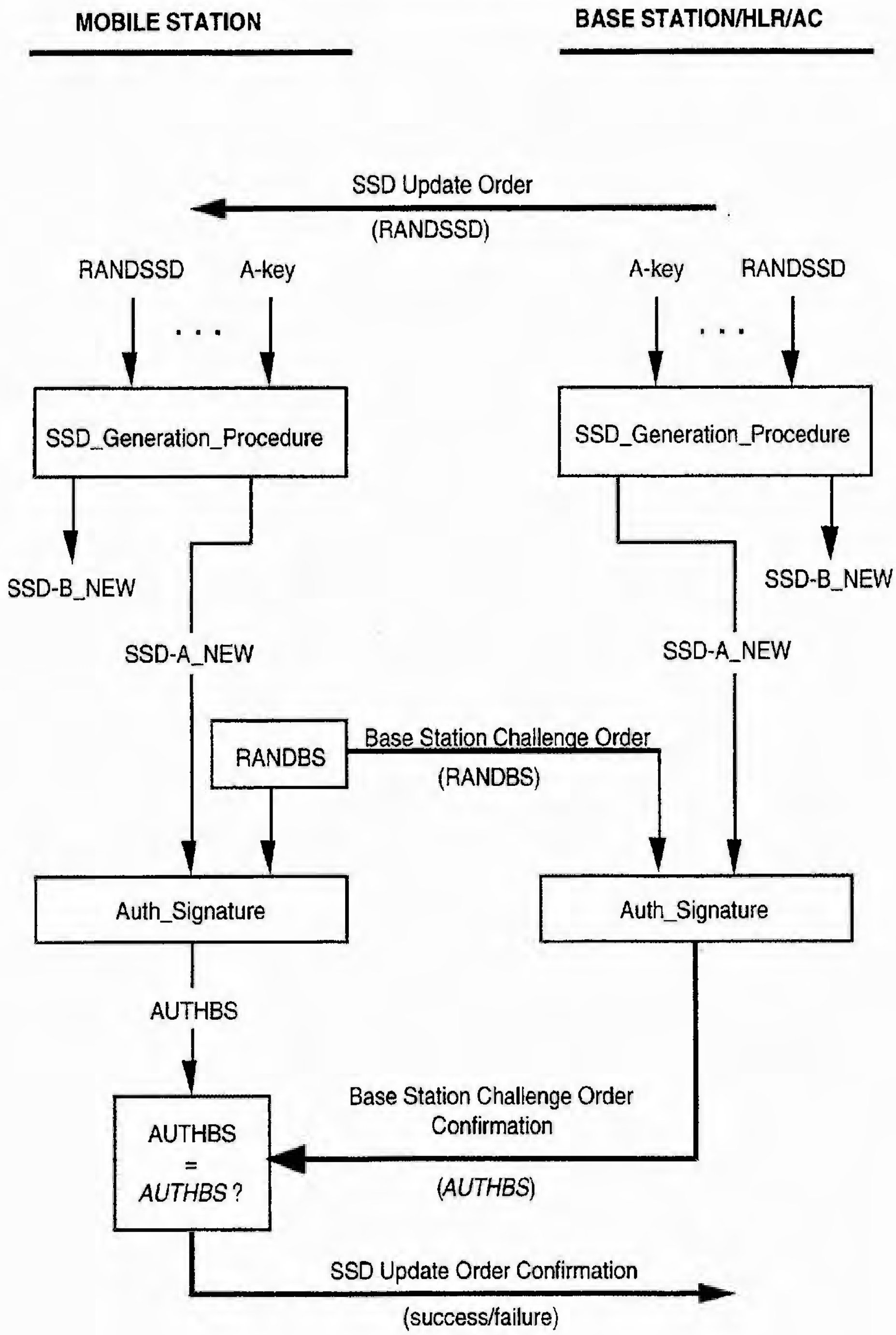
An A-key must be entered into the mobile station. See TSB 50, "User Interface for Authentication Key Entry" for details. In addition to TSB50 procedures, the A-key may be entered into the mobile station via Over-the-Air Activation (see Section 7.2)

More specifically, updating the SSD in the mobile station proceeds as follows (refer to Figure 2.3.12.1.9-1):

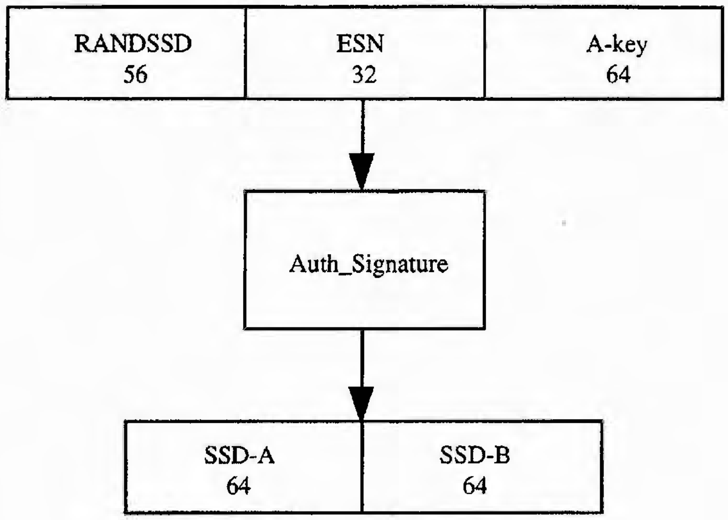
- At the base station:
  - Send an SSD Update Order, with the RANDSSD field set to the same 56-bit random number used in the HLR/AC computations, to the mobile station on the:
    - FOCC in Word 3-First SSD Update Order Word, Word 4-Second SSD-Update Order Word and Word 5-Third SSD Update Order Word of a mobile station control message if the mobile station has not been assigned to an analog voice or digital traffic channel (see Sections 3.6.2.3 and 3.7.1.1); or
    - FDTC in a FACCH message if the mobile station has been assigned to a digital traffic channel (see Sections 3.6.5 and 3.7.3.1.3.2.18); or
    - FVC in Word 2-First SSD Update Order Word and Word 3-Second SSD Update Order Word of a mobile station control message if the mobile station has been assigned to an analog voice channel (see Sections 3.6.4 and 3.7.2.1).

- In the mobile station:
    - Upon receipt of the SSD Update Order, initialize SSD_Generation_Procedure as illustrated in Figure 2.3.12.1.9-2.
    - Execute the SSD_Generation_Procedure algorithm (see Section 2.3.12.1.11).
    - Set SSD-A_NEW equal to the 64 most significant bits of the SSD_Generation_Procedure algorithm output, and SSD-B_NEW to the 64 least significant bits of the SSD_Generation_Procedure algorithm output.
    - Select a 32-bit random number, RANDBS, and send it to the base station in a Base Station Challenge Order on the:
        - RECC in Word C-Base Station Challenge Word if the mobile station is not tuned to an analog voice or digital traffic channel (see Sections 2.6.2.3 and 2.7.1.1); or
        - RDTC in a FACCH message if the mobile station is tuned to a digital traffic channel (see Sections 2.6.5 and 2.7.3.1.3.2.17); or
        - RVC in Words 1 and 2 of a Base Station Challenge Order message if the mobile station is tuned to an analog voice channel (see Sections 2.6.4 and 2.7.2.1).
    - Re-initialize Auth_Signature as illustrated in Figure 2.3.12.1.9-3.
    - Execute the Auth_Signature algorithm (see Section 2.3.12.1.11).
    - Set AUTHBS equal to the 18 bits of the Auth_Signature algorithm output.

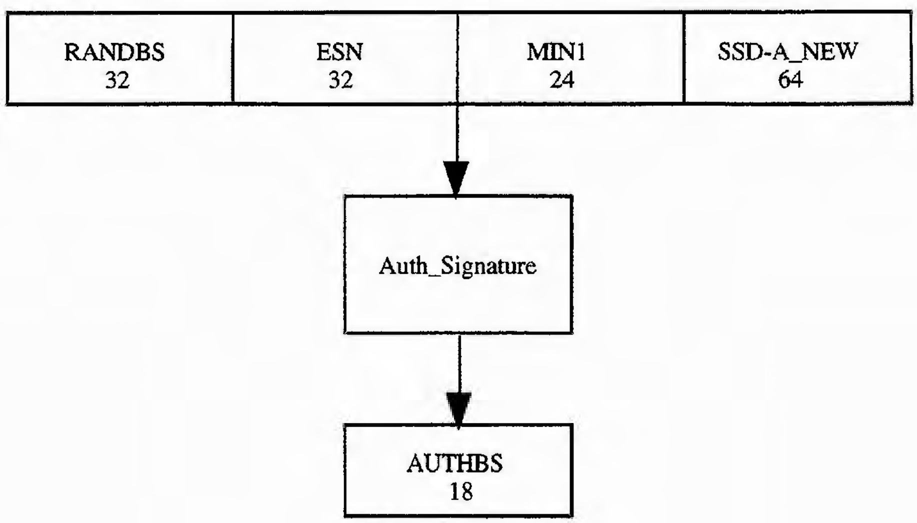
Figure 2.3.12.1.9-1 SSD Update Message Flow



1 **Figure 2.3.12.1.9-2 Computation of Shared Secret Data (SSD)**



2 **Figure 2.3.12.1.9-3 Computation of AUTHBS**



- In the base station:
  - Upon receipt of the Base Station Challenge Order, initialize Auth_Signature as illustrated in Figure 2.3.12.1.9-3, where RANDBS is set to the value received in the Base Station Challenge Order.
  - Execute the Auth_Signature algorithm (see Section 2.3.12.1.11).
  - Set AUTHBS equal to the 18 bits of the Auth_Signature algorithm output.
  - Acknowledge receipt of the Base Station Challenge Order by including AUTHBS in the Base Station Challenge Order Confirmation message, which is sent on the:
    - FOCC in Word 3-Base Station Challenge Order Confirmation Word of a mobile station control message if the mobile station has not yet been assigned to an analog voice or digital traffic channel (see Sections 3.6.2.3, 3.6.3.3 and 3.7.1.1); or
    - FDTC in a FACCH message if the mobile station has been assigned to a digital traffic channel (see Sections 3.6.5 and 3.7.3.1.3.2.19); or
    - FVC in Word 2-Base Station Challenge Order Confirmation of a mobile station control message if the mobile station has been assigned to an analog voice channel (see Sections 3.6.4 and 3.7.2.1).
- In the mobile station:
  - Upon receipt of the Base Station Challenge Order Confirmation, compare the AUTHBS received to that generated internally.
  - Acknowledge receipt of the SSD Update Order as follows:
    - If the comparison at the mobile station is successful, set SSD-A and SSD-B to SSD-A_NEW and SSD-B_NEW, respectively, and:
      - If the mobile station is not tuned to an analog voice or digital traffic channel:
        - Send an order confirmation message to the base station on the RECC with:
          - The "T" field in Word A-Abbreviated Address Word set to 0 to identify the message as an Order Confirmation.
          - The "ORDER" field in Word B-Extended Address Word set to 10101 to signify confirmation of the SSD Update Order.
          - The "ORDQ" field in Word B-Extended Address Word set to 001 to denote the successful completion of the SSD Update process.
          - All other fields set as described in Section 2.7.1.1 and in the references cited therein.
      - If the mobile station is tuned to a digital traffic channel,
        - Send an SSD Update Order Confirmation message to the base station on the RDTC-FACCH with the SSD_UPDATE information element set to 1 and all other parameters set as described in Sections 2.7.3.1.3.2.16 and 2.7.3.1.3.3.

- If the mobile station is tuned to an analog voice channel,
    - Send an Order Confirmation message to the bast station on the RVC with:
        - The "T" field set to 1 to identify the message as an order confirmation.
        - The "ORDER" field set to 10101 to signify confirmation of the SSD Update Order.
        - The "ORDQ" field set to 001 to denote the successful completion of the SSD Update process.
        - All other fields set as described in Section 2.7.2.1 and in the references cited therein.

- If the comparison at the mobile station fails, discard SSD-A_NEW and SSD-B_NEW, and:
    - If the mobile station is not tuned to an analog voice or digital traffic channel,
        - Send an order confirmation message to the base station on the RECC with:
            - The "T" field in Word A-Abbreviated Address Word set to 0 to identify the message as an Order Confirmation.
            - The "ORDER" field in Word B-Extended Address Word set to 10101 to signify confirmation of the SSD Update Order.
            - The "ORDQ" field in Word B-Extended Address Word set to 000 to denote the unsuccessful completion of the SSD Update process.
            - All other fields set as described in Section 2.7.1.1 and in the references cited therein.

- If the mobile station is tuned to a digital traffic channel,
    - Send an SSD Update Order Confirmation message to the base station on the RDTC-FACCH with the SSD_UPDATE information element set to 0 and all other parameters set as described in Sections 2.7.3.1.3.2.16 and 2.7.3.1.3.3.

- If the mobile station is tuned to an analog voice channel,
    - Send an Order Confirmation message to the bast station on the RVC with:
        - The "T" field set to 1 to identify the message as an order confirmation.
        - The "ORDER" field set to 10101 to signify confirmation of the SSD Update Order.

- The "ORDQ" field set to 000 to denote the unsuccessful completion of the SSD Update process.
- All other fields set as described in Section 2.7.2.1 and in the references cited therein.

In the base station, if the SSD Update Confirmation received from the mobile station indicates a success, set SSD-A and SSD-B to the values received from the HLR/AC (see EIA/TIA IS-41).

2.3.12.1.10 Re-Authentication

The Re-Authentication Procedure is initiated by the base station and is performed over a digital traffic channel. This procedure shall be initiated only once during a call, and only during Over-the-Air Activation (see Section 7.2 of IS-136.1).

More specifically:

- At the base station:
  - Generate a 32-bit random pattern referred to as RANDRA (typically a copy of the RAND, see Section 2.3.12.1.2) and send to the mobile station via the FDTC in a Re-Authentication Order FACCH message (see Sections 3.6.5.4 and 3.7.3.1.3.2.30).
- At the mobile station:
  - Set the input parameters to the Auth_Signature Procedure (see Interface Specification for Common Cryptographic Algorithms) as illustrated in Figure 2.3.12.1.10-1, with SAVE_REGISTERS set to TRUE.
  - Execute the Auth_Signature Procedure.
  - Set AUTHRA equal to the 18-bit output of the Auth_Signature Procedure. (Note that the value of AUTHRA is not affected by the value of SAVE_REGISTERS).
  - Send AUTHRA to the base station via the RDTC in a Re-Authentication Order Confirmation FACCH message.

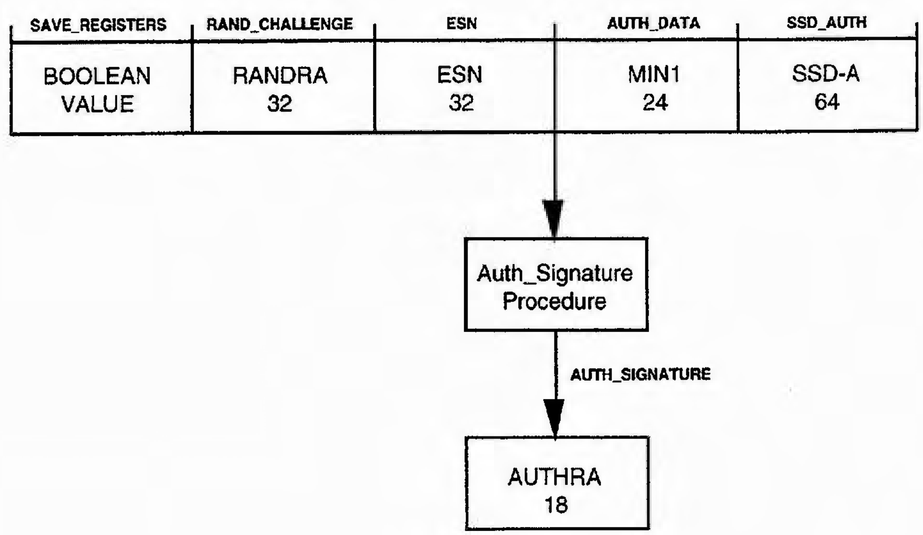
Upon receipt of the Re-Authentication Order Confirmation from the mobile station, the base station shall:

- Set the input parameters to the Auth_Signature Procedure (see Interface Specification for Common Cryptographic Algorithms) as illustrated in Figure 2.3.12.1.10-1, with SAVE_REGISTERS set to TRUE.
- Execute the Auth_Signature Procedure.
- Set AUTHRA equal to the 18-bit output of the Auth_Signature Procedure. (Note that the value of AUTHRA is not affected by the value of SAVE_REGISTERS).
- Compare the value for AUTHRA computed internally with the value of AUTHRA received from the mobile station.

If the comparison by the base station fails, the base station may deny further access attempts by the mobile station, drop the call in progress, or initiate the process of updating the SSD (see Section 2.3.12.1.9).

If the comparison by the base station is successful, this indicates that both the MS and the BS have an identical Voice Privacy Mask and Cellular Message Encryption keys, which are available for subsequent use.

Figure 2.3.12.1.10-1 Computation of AUTHRA for Re-Authentication Procedure



2.3.12.1.11 Auth_Signature, SSD_Update and SSD_Generation_Procedure Algorithm

For details refer to Appendix A. Appendix A, and its associated documents, contain information which is governed under the U.S. International Traffic and Arms Regulation (ITAR) and the Export Administration Regulations. TIA will act as the focal point and facilitator for making such information available. Procedures for distribution of this information are contained in the Technology Transfer Control Plan which is available from TIA.

2.3.12.2 Signaling Message Encryption

In an effort to enhance the authentication process, and to protect sensitive subscriber information, provisions have been made to allow for the encryption of a select subset of FVC, RVC, FDTC and RDTC signaling messages. Note that some fields of the messages subject to encryption are always transmitted as plain text. Order/Message Type fields, for example, are never encrypted.

Consult Appendix A and its associated documents for the list of messages subject to encryption, the details of the encryption algorithm and a description of how the algorithm is initialized and applied. Appendix A, and its associated documents, contain information which is governed under the U.S. International Traffic and Arms Regulation (ITAR) and the Export Administration Regulations. TIA will act as the focal point and facilitator for making such information available. Procedures for distribution of this information are contained in the Technology Transfer Control Plan which is available from TIA.

2.3.12.2.1 Signaling Message Encryption Control

Signaling message encryption is controlled on a per-call basis. The default value is "off". To activate signaling message encryption, the base station must initiate one of the following transactions:

- For mobile stations operating on an Analog Control Channel:
  - on the FOCC,
    - set MEM to 1 in the Initial Digital Traffic Channel Designation message (see Section 3.7.1.1).
- For mobile stations operating on a Digital Control Channel:
  - on the FDCCH,
    - set MEA, MEK, and MED according to the Message Encryption Mode information element sent in the Digital Traffic Channel Designation message.
- For mobile stations assigned to a Digital Traffic Channel:
  - on the FDTC,
    - send a Status Request Message to the mobile station with the MEM field of the Message Encryption Mode A information element set to 1; or
    - hand the mobile station off to a Digital Traffic Channel with the Message Encryption Mode B information element set to 1 in the Handoff message; or
    - hand the mobile station off to a Digital Traffic Channel with MEA, MEK and MED set according to the Message Encryption Mode C information element sent in the Dedicated Digital Traffic Channel Handoff message.
- For mobile stations assigned to an Analog Voice Channel:
  - on the FVC,
    - send a Message Encryption Mode Order with the Order Qualifier field set to 001; or
    - hand the mobile station off to a Digital Traffic Channel with the MEM field set to 1.

Note that regardless of when signaling message encryption is activated, the data used to initialize the algorithm is computed based on parameters in effect at the time the AUTHR appended to the origination/page response message was computed (see Sections 2.3.12.1.7 and 2.3.12.1.8, see also Sections 6.3.12.7 and 6.3.12.8 in IS-136.1).

Once activated, signaling message encryption can be deactivated by the base station as follows:

- For mobile stations assigned to Digital Traffic Channels:
    - on the FDTC,
        - send a Status Request Message to the mobile station with the MEM field of the Message Encryption Mode A information element set to 0; or
        - hand the mobile station off to a Digital Traffic Channel with the Message Encryption Mode B information element set to 0 in the Handoff message; or
        - hand the mobile station off to a Digital Traffic Channel with MEA, MEK and MED set to 0 in the Message Encryption Mode C information element sent in the Dedicated Digital Traffic Channel Handoff message.
- For mobile stations assigned to an Analog Voice Channel:
    - on the FVC,
        - send a Message Encryption Mode Order with the Order Qualifier field set to 000; or
        - hand the mobile station off to a Digital Traffic Channel with the MEM field set to 0.

In all cases both the base station and mobile station shall continue to operate in their present mode until the message sent to the mobile station has been properly acknowledged.

2.3.12.3 Voice Privacy

The term Voice Privacy refers to the process by which user voice transmitted over a digital traffic channel is afforded a modest degree of cryptographic protection against eavesdropping in the mobile station - base station segment of the connection.

Note that regardless of when voice privacy is activated, the data used to initialize the algorithm is computed based on parameters in effect at the time the AUTHR appended to the origination/page response messages was computed (see Sections 2.3.12.1.7 and 2.3.1.12.1.8, see also Sections 6.3.12.7 and 6.3.12.8 in IS-136.1).

2.3.12.3.1 Voice Privacy Control

Requests to activate/deactivate the voice privacy feature may be made during the call setup process or while the mobile station is in the conversation state. In either case, however, the decision to honor the request lies with the base station. Furthermore, the mobile station must not act under the assumption that the request has been granted until it receives positive verification from the base station.

2.3.12.3.1.1 Voice Privacy Control During Call Establishment

2.3.12.3.1.1.1 Mobile Station Originations

2.3.12.3.1.1.1.1 Analog Operation

To request activation of voice privacy on mobile station originations, a digital-privacy specification is included in Word B of the RECC Origination message (See Section 2.3.12.3).

Voice privacy is activated in the mobile station only when the PM field in the First Digital Channel Assignment Word (see Section 3.7.1) is set to 1.

2.3.12.3.1.1.1.2 Digital Operation

The Voice Mode information element is included with a mobile station Origination in order to request activation of voice privacy.

The mobile station activates voice privacy upon reception of a Digital Traffic Channel Designation containing a Voice Mode information element.

The form of voice privacy to be activated is identified in the Voice Mode information element. If no Voice Mode information element is included in the Digital Traffic Channel Designation, the mobile station shall not enable voice privacy.

2.3.12.3.1.1.2 Mobile Station Terminations

2.3.12.3.1.1.2.1 Analog Operation

To request activation of voice privacy on mobile station terminations, a digital-privacy specification is included in Word B of the RECC Page Response message (See Section 2.3.12.3).

Voice privacy is activated in the mobile station only when the PM field in the First Digital Channel Assignment Word (see Section 3.7.1) is set to 1.

2.3.12.3.1.1.2.2 Digital Operation

The Voice Mode information element is included with a mobile station Page Response in order to request activation of voice privacy.

The mobile station activates voice privacy upon reception of a Digital Traffic Channel Designation containing a Voice Mode information element.

The form of voice privacy to be activated is identified in the Voice Mode information element. If no Voice Mode information element is included in the Digital Traffic Channel Designation, the mobile station shall not enable voice privacy.

2.3.12.3.1.2 Voice Privacy Control After Initial Channel Assignment

To request a change in the privacy mode after the mobile station has been assigned to a Digital Traffic Channel, a Status message with the Service Privacy Mode A information element set to the requested value (0 = privacy off, 1 = privacy on) is sent to the base station on the RDTC.

The base station acknowledges this Status message by returning a Base Station Ack message with the Message Type parameter set to Status. The mobile station does not change its privacy mode as a result of receiving this Base Station Ack message. The mobile station shall only change its privacy mode if, after receiving a Base Station Ack message, it subsequently receives a Status Request message from the base station (see Sections 2.6.5.3.2 and 2.6.5.4).

To request voice privacy after the mobile station has been assigned to an Analog Voice Channel, a Page Response message with the Order Qualifier field set to the 100 and the Message Type field set to XXX1X or XX1XX is sent to the base station on the RVC. The mobile station continues to operate in its current mode until it receives the corresponding Call Mode Ack message on the FVC and is subsequently handed off to a Digital Traffic Channel where the PM field in the First Digital Channel Assignment Word has been set to 1.

2.3.12.3.2 Cipher Placement

Enciphering shall take place after error correction coding and before interleaving. In particular, note that user voice is enciphered while still represented as bits rather than quaternary symbols. Similarly, deciphering occurs after deinterleaving.

2.3.12.3.3 Voice Privacy Algorithm

For details refer to Appendix A. Appendix A, and its associated documents, contain information which is governed under the U.S. International Traffic and Arms Regulation (ITAR) and the Export Administration Regulations. TIA will act as the focal point and facilitator for making such information available. Procedures for distribution of this information are contained in the Technology Transfer Control Plan which is available from TIA.

2.3.12.4 Data Privacy

The term Data Privacy refers to the process by which user data transmitted over a digital traffic channel is afforded a modest degree of cryptographic protection against eavesdropping in the mobile station - base station segment of the connection.

2.3.12.4.1 Data Privacy Control

Requests to activate or deactivate the data privacy feature may be made during the call setup process. The decision to honor the request lies with the base station. The mobile station shall not act under the assumption that the request has been granted until it receives verification from the base station (i.e., a digital traffic channel assignment).

2.3.12.4.2 Mobile Station Originations

2.3.12.4.2.1 Analog Operation

To request activation of data privacy on mobile station originations, a digital-privacy specification is included in Service Code Word D of the RECC Origination with Service message. The voice-privacy mode indicated in the MSB of the Order Qualifier shall be set to 0.

Data privacy is activated in the mobile station only when the PM field in the First Digital Channel Assignment Word (see Section 3.7.1) is set to 1.

2.3.12.4.2.2 Digital Operation

The Data Mode information element is included with a mobile station Origination in order to request activation of data privacy.

The mobile station activates data privacy upon reception of a DTC Designation message.

The form of data privacy to be activated is identified in the Data Mode information element. If no Data Mode information element is included, the mobile station shall not enable data privacy.

2.3.12.4.3 Mobile Station Terminations

2.3.12.4.3.1 Analog Operation

To request activation of data privacy on mobile station terminations, a digital-privacy specification is included in Service Code Word D of the RECC Page Response with Service message. The voice-privacy mode indicated in the MSB of the Order Qualifier shall be set to 0.

Data privacy is activated in the mobile station only when the PM field in the First Digital Channel Assignment Word (see Section 3.7.1) is set to 1.

2.3.12.4.3.2 Digital Operation

The Data Mode information element is included with a mobile station Page Response in order to request activation of data privacy.

The mobile station activates data privacy upon reception of a DTC Designation message.

The form of data privacy to be activated is identified in the Data Mode information element. If no Data Mode information element is included, the mobile station shall not enable data privacy.

2.3.12.4.4 Data Privacy Control After Initial Channel Assignment

Mobile stations and base stations shall not request a change in data privacy after initial channel assignment.

2.3.12.4.5 Data Privacy Algorithm

The availability of this information is governed under the U.S. International Traffic and Arms Regulation (ITAR) and the Export Administration Regulations. Refer to IS-130.

2.4 Supervision

For supervising the connection on the traffic channel

- if the mobile station is on an analog voice channel the Supervisory Audio Tone (see Section 2.4.1) and the Signaling Tone (see Section 2.4.2) are used.
- if the mobile station is on a digital traffic channel the Digital Verification Color Code (see Section 2.4.3) is used.

2.4.1 Supervisory Audio Tone

The supervisory audio tone (SAT) shall be one of three frequencies: 5970, 6000, or 6030 Hz. The SAT is added to the voice transmission by a base station (see Section 3.4.1). A mobile station must detect, filter, and modulate the transmitted voice channel carrier with this tone. Transmission of the SAT by a mobile station must be suspended during transmission of wideband data on the reverse voice channel (see Section 2.7.2), but must not be suspended when signaling tone is sent (see Section 2.4.2).

While a valid SAT is detected and the measured SAT determination does not agree with the SAT color code (SCCr) received in the mobile station control message (see Sections 3.7.1.1 and 3.7.2), the receiver audio must be muted.