DIGITAL CHIP SET BRINGS SUPERIOR RANGE, VOICE CLARITY, AND NOISE IMMUNITY TO CORDLESS TELEPHONY

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INTRODUCTION

A new digital chip set developed jointly by Zilog Incorporated in Campbell, California and LinCom Corporation in Los Angeles, California promises to usher in a new generation of cordless telephones which offer vastly improved performance over existing cordless phones.

The availability of frequency spectrum in the 902 to 928 MHz band, also known as the ISM band, for unlicensed operation has spurred the development of a new generation of wireless products for applications ranging from wireless local area networks, point of sale terminals, to point-to-point radios in addition to digital cordless telephones.

The Z87000 and the Z87010 represent a two chip solution which provides mixed mode digital/analog circuitry to perform the majority of the signal processing functions between IF and baseband frequencies which are required for cordless telephone operation. The two chip set is bundled into a package providing a complete reference solution for OEM vendors which includes the man machine interface and telco support software, RF circuit schematics, and product development platforms.

CORDLESS TELEPHONY

Cordless telephones were originally introduced to the mass consumer market in the 1980's. These first generation phones transmit and receive in the 46 and 49 MHz bands in the United States. Employing principally analog technology, they use FM modulation to relay full duplex voice between the handset and basestation. The low cost and complexity of these phones has enabled affordability and in turn resulted in widespread consumer acceptance. Nevertheless, phone performance is highly susceptible to background noise and other types of man-made interference. Furthermore, range is extremely limited due to multipath fading and other signal attenuation caused by interior walls and structures typical to household environments. In addition, co-existence between multiple cordless phones in close proximity is severely restricted due to co-channel interference. Indeed, many consumers frequently complain about hearing their neighbor's phone conversations or being altogether unable to place or receive calls. These analog phones have been refined in recent years to include the use of multiple frequency channels in order to avoid interference. Current phones provide 10-channel capability, some with automatic channel selection. The FCC recently approved 25 channel phone operation. However, such improvements only marginally address the major performance impediments such as multipath and co-channel

interference which usually limits phone operating range and can especially degrade analog voice quality.

In 1985, however, the U.S. Federal Communications Commission drafted the Part 15.247 regulations permitting unlicensed use of three frequency bands for Industrial, Scientific, and Medical applications (denoted as the ISM bands). The three bands include spectrum between 902 to 928 MHz, 2400 to 2483.5 MHz, and 5725 to 5850 MHz. The FCC regulations allow considerable design flexibility to address a wide array of voice and data communication applications. The use of the 900 MHz ISM band for cordless telephony inherently provides the possibility of improved range because of the better propagation characteristics compared to 46/49 MHz. In addition, the FCC regulations allow up to 1 Watt of transmit power which will further increase range and improve communications performance. The availability of great bandwidth at 900 MHz than at 46/49 MHz makes the use of digital transmission and reception more easily implementable. Although this band is not exclusively set aside for spread spectrum operation, applications employing spread spectrum techniques will enjoy improved interference immunity, increased range, inherent privacy and security, and greatly enhanced voice quality.

Currently between 18 to 20 million cordless phones are sold annually in the United States (according to the EIA). Almost 52% of all US households possess a cordless telephone which represents a significant penetration of the US population. The overwhelming majority of these are analog phones operating at 46/49 MHz. However, it is estimated that by the end of 1995, almost 5 to 10% of all cordless phones sold will be 900 MHz phones. The market share is expected to increase as enabling technology is developed to reduce the retail price point to well below $200.00. Over the last two years, a number of new cordless phone products have been introduced using the 900 MHz ISM band. These employ either completely analog transmission or digital transmission. However, until now, product offerings employing digital spread spectrum transmission have been somewhat limited and have fallen short of the expectations of the technology.

The key challenges to designing cordless telephones operating at 900 MHz have been reaching a reasonable cost and battery life comparable to the 46/49 MHz phones. In meeting these challenges, Zilog has developed its ZPhone cordless phone system design upon the Z87000 spread spectrum controller and Z87010 ADPCM processor. While other chip sets developed for use in spread spectrum cordless telephones are available, Zilog's Z87000/Z87010 chip set combination is the only complete design solution available. As part of its effort to support cordless phone manufacturers in developing 900 MHz cordless telephones, Zilog not only provides the IC's which perform all of the IF to baseband signal processing, but it also provides source code and object libraries for software residing on an integrated processor which performs the complete communications protocol and supports all man machine interface and telco functions, complete circuit schematics including a reference RF section design, a comprehensive bill of materials, full documentation and technical support.

ZPHONE SYSTEM OVERVIEW

Zilog's Z87000 Cordless Phone Transceiver/Controller and Z87010 ADPCM Processor together allow implementation of a 900 MHz spread spectrum phone that is in compliance with the United States Federal Communications Commission regulations for operation in the ISM band. These chips are designed to support a specific phone system design—the "ZPhone"—which uses frequency hopping and digital frequency shift keying (FSK) modulation to permit high power transmission and extended range, 32 kbps ADPCM speech coding to achieve high voice quality, adaptive frequency hopping to reduce the effects of in-band interferers, adaptive transmit power control to extend battery life, and an overall system approach that minimizes system costs.

The spread spectrum technique employed is frequency hopping. The hop dwell time is 4 milliseconds. Data is communicated between the handset and basestation via time division duplex (TDD) operation. Each frequency dwell time defines a single TDD frame slot. These TDD frame slots are in turn divided into a time slot during which the basestation transmits a data packed to the handset and a time slot during which the handset transmits a data packet to the basestation. These transmits slots are separated by two distinct guard time slots. Figure 1 depicts the TDD frame structure. The guard times between signal transmission and reception and between frequency hops are provided to allow time for hardware switching and settling. Frame timing is generated via frame counters at the handset and basestation. The basestation establishes the time reference for the overall ZPhone system. The handset tracks the basestation frame timing. The difference in values between the basestation and handset frame counters thus reflect the propagation delay between the basestation and the handset.
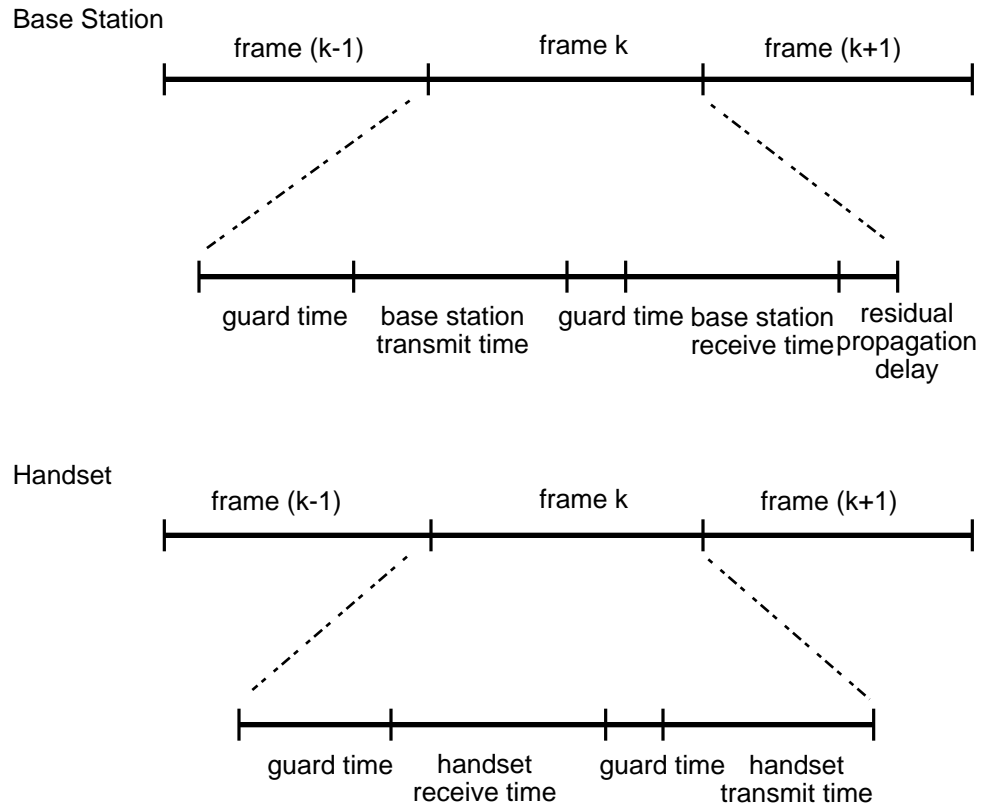
Base Station

frame (k-1)   frame k   frame (k+1)

guard time   base station   guard time base station   residual
              transmit time   receive time propagation
                                                            delay

Handset

frame (k-1)   frame k   frame (k+1)

guard time   handset   guard time   handset
               receive time   transmit time

Figure 1. TDD Frame Structure

The burst transmission rate is 93 kbps. The 3 dB transmission bandwidth for each hop channel is 180 kHz. With a channel separation of 182 kHz, a total of 142 possible hop frequencies are available between 902 MHz and 928 MHz. At any given time, only 64 of these channels are actively used. The ZPhone system is designed to adaptively avoid those channels with severe interference. This is accomplished by replacing active frequencies which possess poor SNR conditions with hop frequencies from the pool of remaining inactive frequencies. The hopping sequence, i.e. the order in which the active frequencies are hopped to, is generated according to the state of a pseudo-noise sequence generator.

Table 1. Signaling-Multiplex (S-MUX) Format

64 bits	16 bits	8 bits	24 bits	16 bits	16 bits	4 bits
preamble	unique word	code seed	ID security code	control	signaling	antenna switching (handset only)

Table 2. Talk Multiplex (T-Mux) Format

128 bits	16 bits	4 bits
voice	signaling (handset only)	antenna switching

Data is packetized and transmitted in burst during each TDD frame in either Signaling Multiplex (S-Mux) or Talk Multiplex (T-Mux) frame formats. At the basestation, the transmit packets are 144 bits; at the handset the transmit packet is 148 bits. The S-Mux format is used to establish two-way link communication between the handset and basestation. It is optimized for fast acquisition and synchronization. The T-Mux format is used during voice communications between the handset and basestation. Tables 1 and 2 depict the S-Mux and T-Mux frame formats. The S-Mux frame consists of a 64-bit alternating 1 and 0 preamble. This preamble is followed by a 16-bit unique word sequence. After the unique word is an 8-bit field which is updated each frame to represent the state of the pseudo-noise generator register which is used to determine the hopping sequence. The code seed field is followed by a 24-bit ID security code field. This ID code is used to prevent unauthorized access to the cordless phone system. The initial value of the ID security code is set to a default value upon power up but is randomized as time progresses so that a handset can only access its own designated basestation and vice versa. The ID security code is followed by a 16-bit control field. This control field is used to transfer basic alert and link setup information. Finally the last 16-bits are reserved for the transfer of signaling information. For the handset, the transmit packed is 4 bits longer because of the addition of a fixed known pattern used in the antenna diversity switching algorithm.

The basestation hardware can be configured to operate with dual antenna ports to provide antenna diversity. At the end of the reception of a frame from the handset, the basestation performs an energy measurement at each antenna port during the last four bits which are always known. On the basis of these measurements, the antenna switching is toggled to gate the antenna port with the largest energy measurement from the previous frame during the next frame.

The T-Mux format consists of 128 bits of voice data corresponding to 32 4-bit voice sample nibbles. The T-Mux frame also has 16-bit field dedicated for signaling and, for the handset, contains an additional four bits of a fixed pattern to be used for antenna diversity. In order to ensure sufficient data transitions to support accurate symbol timing synchronization, the polarity of several of the bits in the voice field is reversed.

At the handset there are four operating modes: Acquisition Mode, Wake/Sleep Mode, Talk Mode, and Cradle Mode. At the basestation, there are three operating modes: Idle Mode, Talk Mode and Cradle Mode. Figures 2 and 3 illustrate these modes. Transitions between the modes at the handset are coordinated with the transitions at the basestation and vice versa. The basestation' s Idle Mode corresponds to either the Acquisition Mode or the Wake/Sleep Mode of the handset; the basestation's Talk mode corresponds to the handset's Talk Mode; and the basestation's Cradle Mode corresponds to the handset's Cradle Mode.
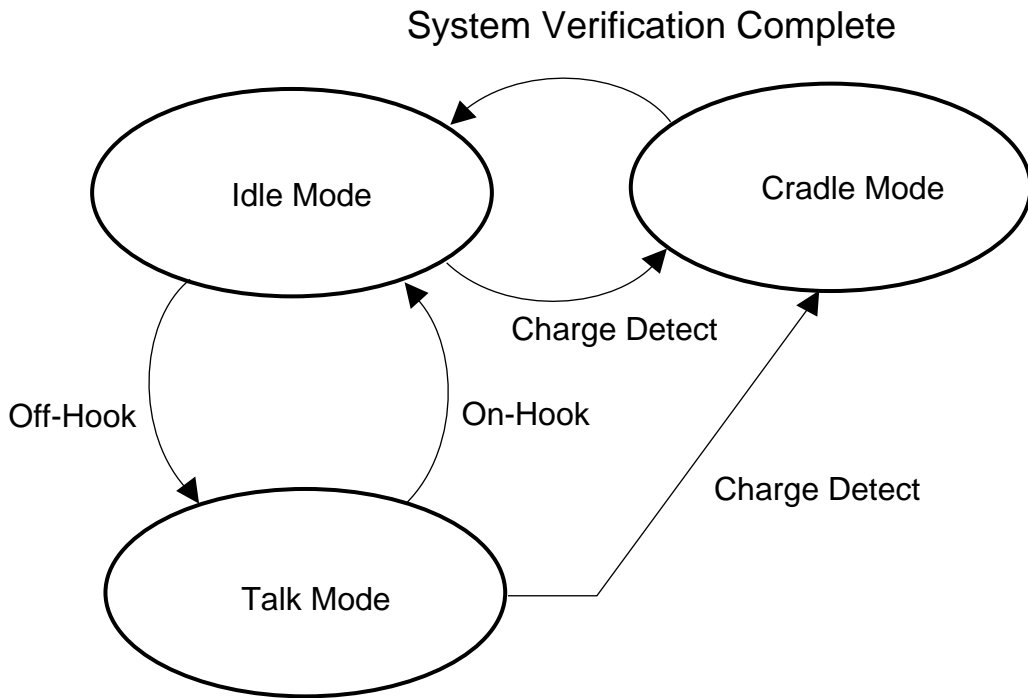
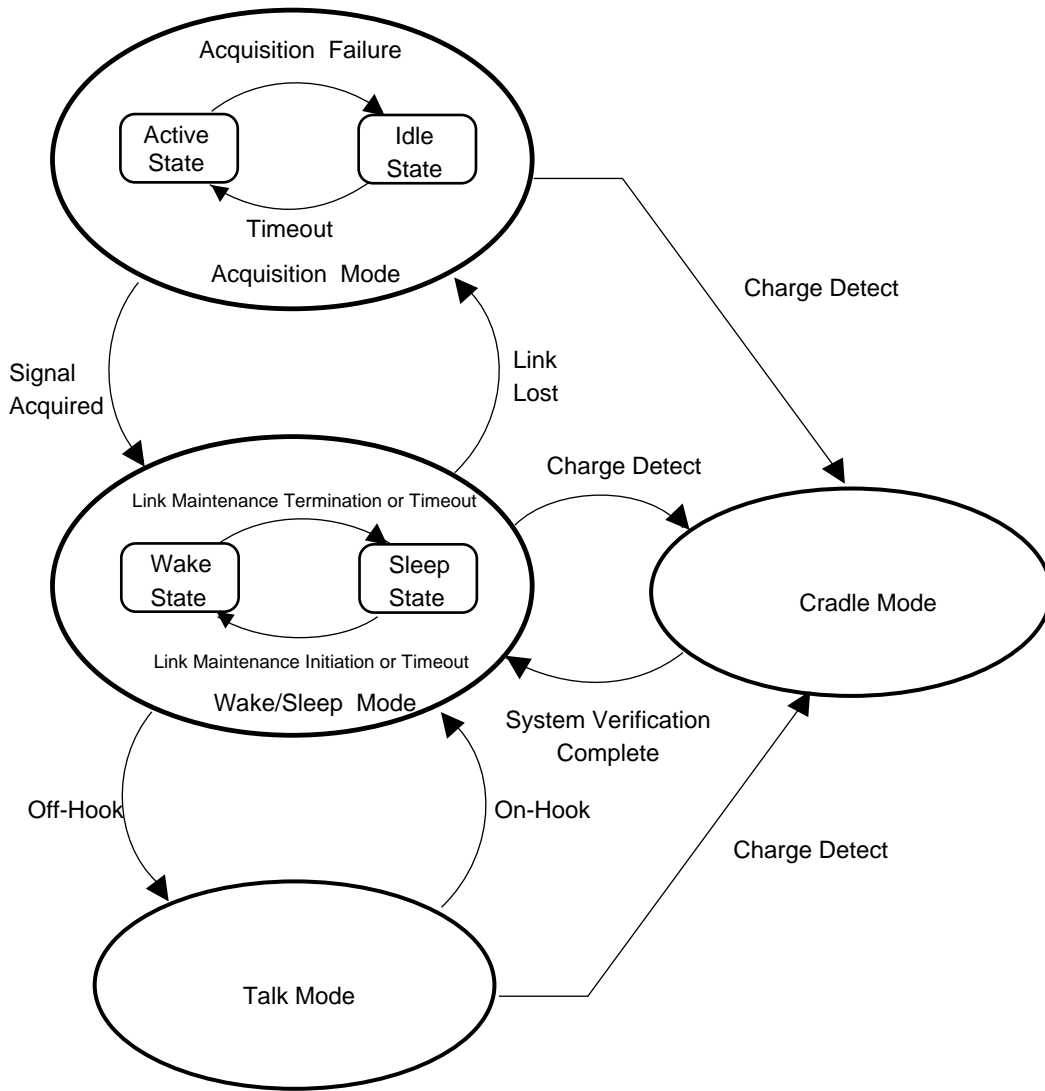**Figure 2. Basestation Operating Modes**

Figure 3. Handset Operating Modes

In Idle Mode, the basestation transmits S-Mux frames for the purpose of handset acquisition when required. The handset remains in a listen only state in Acquisition Mode while attempting to acquire and synchronize its local frame timing to the received S-Mux frames. After synchronizing to the basestation. the handset reverts to Wake/Sleep Mode in which it turns off all power to conserve standby battery life and periodically power on only to ensure that it is still synchronized to the basestation. When a two-way link is required for voice communication, both the handset and basestation activate their Talk modes. The Cradle Mode is activated when the handset is placed in the basestation charger. During this mode the handset and basestation exchange information to verify ID security codes and check for consistency between their frequency hopping tables.

ZPHONE SYSTEM COMPONENTS

Figure 4 depicts the major system components and the signal flow for the ZPhone system. The basestation and handsets are similar in that they both consist of frequency agile RF sections, a Z87000 spread spectrum transceiver controller, a Z87010 ADPCM speech processor, and a PCM Codec interface. The basestation and handset differ primarily in that at the handset the PCM codec interfaces directly to speaker/microphone pair whereas in the basestation, the PCM codec connects to a telephone line interface. Other miscellaneous functions such as the keypad, LED's ringer, et cetera comprise the man-machine interface functions.
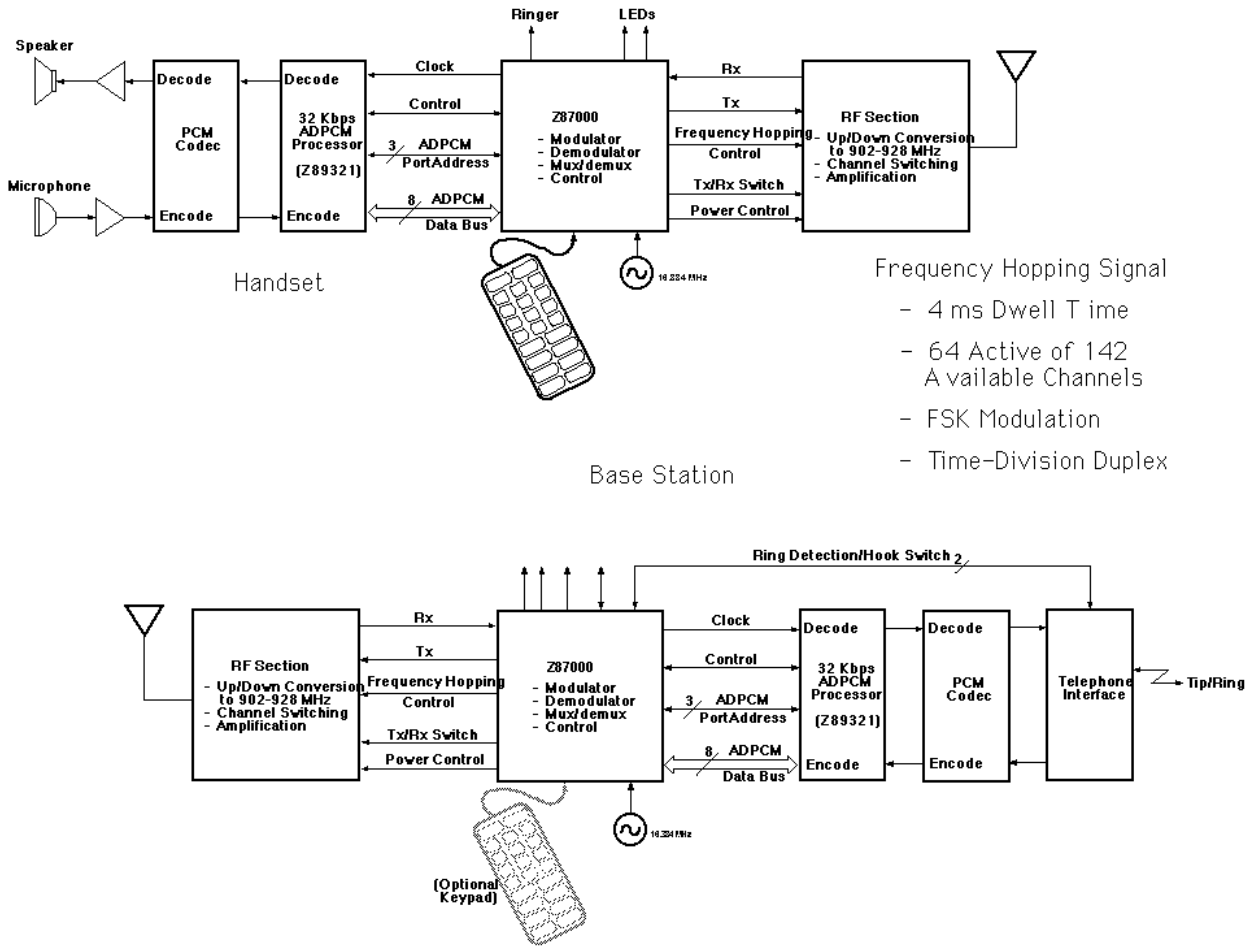

Figure 4. Z87000 Cordless Phone Transceiver/Controller

The RF sections perform amplification, channel switching, upconversion and downconversion to/from 902-928 MHz. Note that the chip set solution is essentially independent of the actual transmitted frequency. Hence, with the appropriate RF design and supporting software, the chip set can be used to implement a cordless telephone operating at the 2.4 or 5.7 GHz ISM bands or any other band, hopping or not hopping. The RF receive and transmit signals interface directly to the Z87000 spread spectrum transceiver controller at an IF frequency of 10.7 MHz.

Figure 5 depicts the Z87000 functional block diagram. For both transmit and receive, the interface signals at the IF are FSK modulated. The chip contains a 1-bit analog to digital converter with oversampling to digitize and process the received IF signal at 10.7 MHz. On-chip digital circuitry performs downconversion, frequency tracking, limiter-discriminator detection, bit synchronization, frame synchronization, and channel signal to noise ratio estimation. The received serial data stream is collected into 4-bit nibbles at a receiver rate buffer. These nibbles are then transferred via an 8-bit bidirectional interface to the voice processor. On the transmit side, a rate buffer collects 4-bit nibble samples from the same 8-bit bidirectional interface. These samples are then converted to serial format processed by digital circuitry which performs FSK modulation and up-conversion to an IF of 10.7 MHz. An on-chip 4-bit digital to analog converter converts the samples from the modulator to provide analog samples for transmission at the RF interface. A separate block contains frame counters and event triggered circuitry to maintain frame timing and generate critical internal and external timing signals. This block also possesses an 8-bit analog to digital converter to sample the analog received signal strength indicator from the RF section and a 4-bit digital to analog converter to provide 16 level control of the transmitted power output.
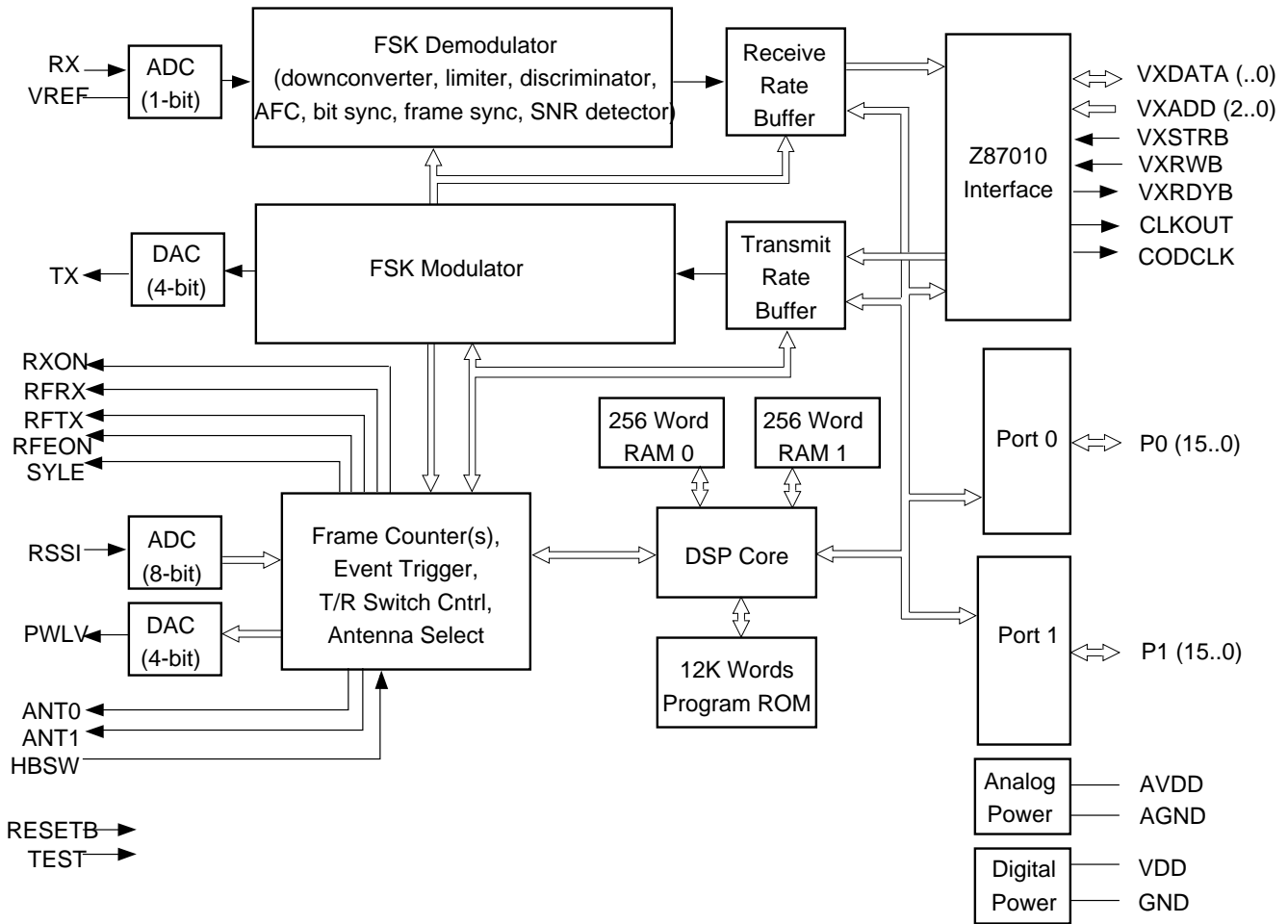
Figure 5. Z87000 Functional Block Diagram

The Z87000 contains a 16-bit fixed point digital signal processing core. The core controls 32 bits of general purpose I/O which can be individually programmed for input or output to support man-machine interface and other telco support functions.

The Z87010 implements the voice compression and decompression algorithms using the CCITT defined G.721 standard for adaptive delta pulse coded modulation. The Z87010 employs essentially the same 16-bit fixed point digital signal processing core as does the Z87000.

ZPHONE LAYERED COMMUNICATION PROTOCOL

Communication between the handset and basestation is performed via a software protocol consisting of three network layers as depicted in Figure 6. Each network layer establishes essentially seamless communication between the handset and basestations. Messages generated at one layer of the basestation or handset are communicated to the corresponding layer at the handset or basestation by first passing through any lower layers and then through the physical radio transmission channel, with the exchange of messages governed by the rules and procedures of that layer's protocol. The network architecture is broadly based on the layered philosophy of the ISO Open Systems Interconnection Reference Manual.
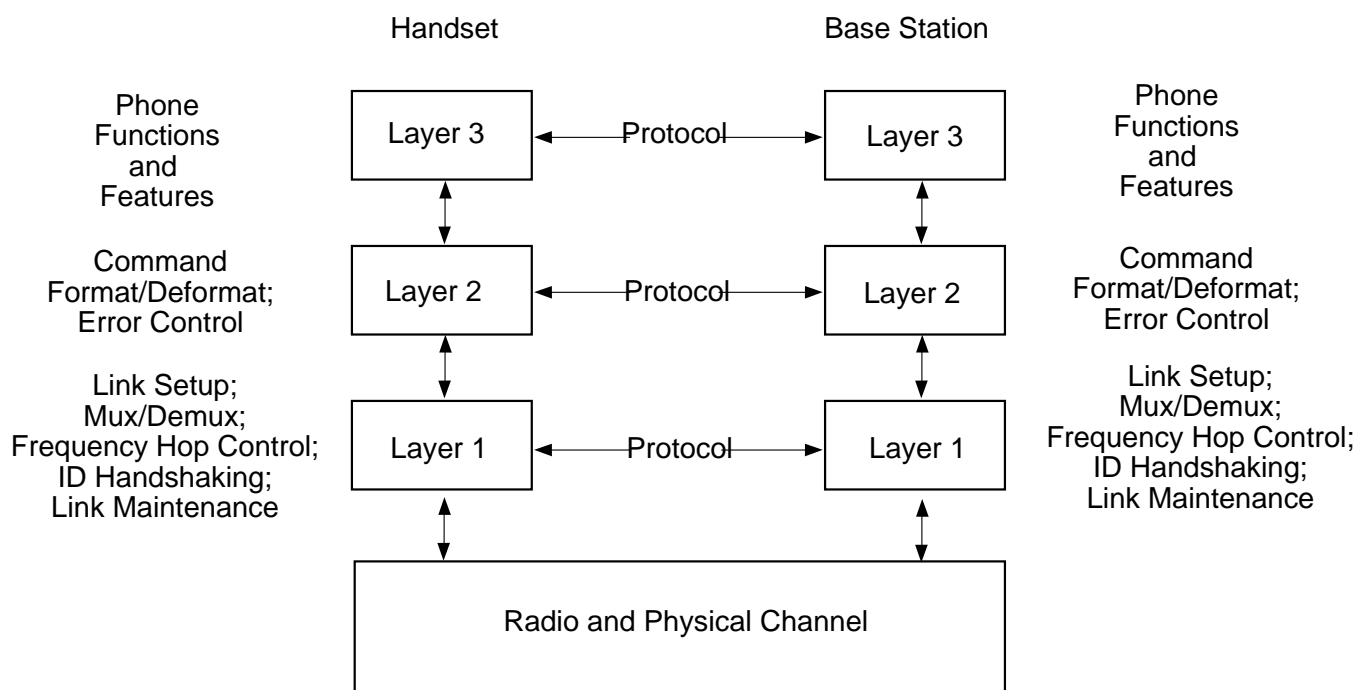
Handset   Base Station

Phone Functions and Features — Layer 3 ←Protocol→ Layer 3 — Phone Functions and Features

Command Format/Deformat; Error Control — Layer 2 ←Protocol→ Layer 2 — Command Format/Deformat; Error Control

Link Setup; Mux/Demux; Frequency Hop Control; ID Handshaking; Link Maintenance — Layer 1 ←Protocol→ Layer 1 — Link Setup; Mux/Demux; Frequency Hop Control; ID Handshaking; Link Maintenance

Radio and Physical Channel

Figure 6. Signaling Architecture

The layer 1 protocol is responsible for operational control of the ZPhone communications circuitry, including support and control of the time division duplexing, signal acquisition, frequency hopping and wake/sleep cycles. Layer 1 also performs multiplexing/demultiplexing of signaling data, ID security code hand-shaking, two-way link communication setup and control, and various link maintenance functions. Link maintenance includes adaptive hop channel frequency assignment, transmit power control, and frequency uncertainty compensation.

The layer 2 protocol is responsible for message formatting of the signaling data, performing acknowledged and unacknowledged signaling data transfer, and signaling data error detection and correction through re-transmission. These layer 2 functions are employed by both the layer 1 and layer 3 protocols to support operation of the Zphone.

Information is shuffled between the handset and basestation using variable length Information Elements. These information elements are broken down and formatted into 64-bit message code words. Message codewords include a control octet indicating the message type and sequence number as well as 2 cyclic redundancy code (CRC) octets for error detection.

The layer 3 protocol uses the facilities provided by layers 1 and 2 to offer support of the man machine interface, support of the interface to the public switched telephone network (PSTN), and support of the interface between the Z87000 itself and the Z87010 voice processor. A layer 3 message is defined in terms of information elements which are delivered by layer 2 to or from the remote end. This "application" layer is typically where user specific functions such as memory dialing, LCD support, et cetera are imple-

mented. The layer 3 software is written entirely in C code for easy maintenance and modification. As part of its reference design solution, Zilog provides fully functional layer 3 software to OEM developers which implement advanced phone features taking advantage of most of the Zphone's system capabilities. However, the source code for this software is provided as well as a firmware development platform which permits manufacturers the ability to customize their own man machine interface.

THE ZPHONE DEVELOPMENT KIT

For faster easier development, Zilog offers reference design development kits. The development kit contains a pair of ZPhone Handset/Basestation Evaluation Boards, ZPhone Software, Z87000 In-Circuit Emulators, Z87000 Protopacks, and the Z87000 Software Development Toolset including a C Compiler and Assembler.

The ZPhone Handset/Basestation Evaluation Board can be configured as either a handset or basestation. It consists of an assembled circuit board, the Z87000 spread spectrum controller with telephone software, and the Z87010 speech encoder/decoder. Also included are a telephone line interface, handset speaker, microphone, and an AC adapter. Two such boards can be connected to RF modules and form a complete, full featured cordless telephone with intercom capabilities. The boards may also be connected through an IF link only and used with the Z87000 In-Circuit Emulator to develop custom application software.

The entire ZPhone software suite is mapped to 12K Words of ROM and provided as part of the reference design development kit. The lower 8K Words make up the lower level software functions which control basic transceiver operation. The upper 4K Words implement application level software including the implementation of special features. The software currently provides the following features:

- Tone/Pulse Dialing

- 20 Number Memory

- Redial/Pause

- Flash

- Hold

- Mute

- Page/Intercom

- Handset Volume Control

- Ringer Volume Control

- Talk, Low Battery, Mute, Charge Indicators

The Z87000 In-Circuit Emulator allows ZPhone OEMs to customize the phone application software to their own specific needs. With the Z87000 ICEBOX, the software can be downloaded from a PC and debuffed in a real time development platform. With this tool, the execution of the code can easily be traced.

The Z87000 Protopack is a simple emulation module which allows the application software to be loaded on EPROM and tested in an application system.

The Z87000 Software Development Toolset consists of an assembler and compiler packages written to assist rapid application code development and debug.

Production samples are currently available. For pricing and delivery information or to obtain sample devices and reference design development kits.