1 Introduction

1.1 Historical Background

Mobile communications systems require a significant financial investment to obtain radio spectrum, which consists of small, but expensive, frequency bands that are used to extend the networks over wide geographic areas. There are additional costs to operate and maintain those networks. Roaming agreements made between service providers can complement insufficient network coverage, but financial constraints still dictate that existing assets such as the backbone networks be re-used whenever possible. As a result, new mobile communications systems are rarely designed without incorporating some elements of earlier systems.

Before discussing the signal processing procedures used by the second and later generations of digital mobile communications systems, it is appropriate to describe the goals and define the performance criteria that were used to construct those procedures. Then we outline the key features of their precursors, the first generation analog mobile communications systems, which introduced the cellular concept. It will become apparent that the design of these early analog systems and the experience gained from operating them had a profound impact on the design of later systems. A more detailed discussion of the technical and social background that drove the development of early mobile communications systems can be found in [Lee (1995), Rappaport (2002)].

1.1.1 Problem Description

In the discussions in this text we have divided the signal processing operations in the mobile communications system into two subsystems: the speech signal processing subsystem and the radio signal processing subsystem. The former incorporates bandwidth-limiting, sampling, and encoding the speech waveform into as few bits as possible while maintaining acceptable speech quality. The latter is concerned with protecting those bits, packaging them, and transmitting them through the network. In some sense the distinction is artificial since the two subsystems interact and are typically implemented on the same processors. On the other hand, they were typically developed by researchers with different technical backgrounds and in most cases are defined by different standards or different parts of the same standard. Whenever we use the term signal processing operations, this should be understood to mean both subsystems taken together.

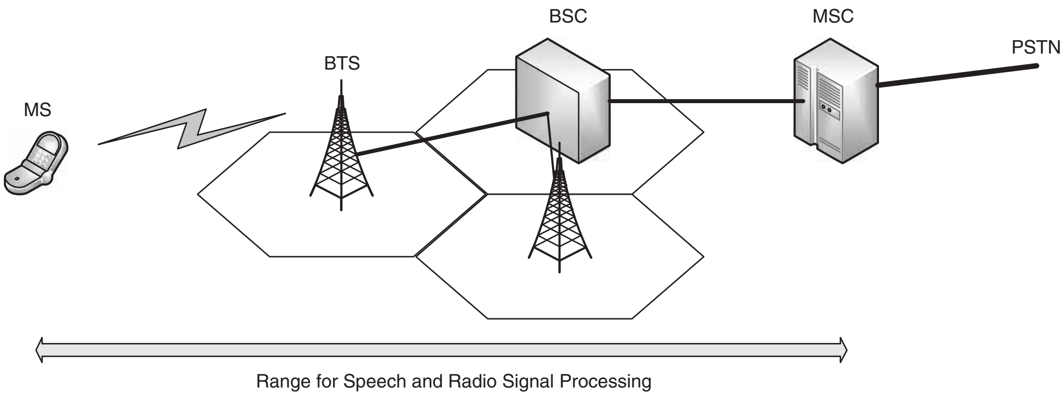

Fig. 1.1 Network architecture of circuit-switched mobile communications systems.

Figure 1.1 shows a generic network architecture that represents the signal processing operations employed by the second generation digital circuit-switched mobile communications systems. In this architecture, once a call is established, the Mobile Station (MS) transforms a short, typically 20 ms, segment of speech into an appropriate digital format, and then transmits it to one or more Base Transceiver Stations (BTS). During the end-to-end transmission from the MS to the far-end device, which may be another MS or a fixed telephone, the speech is represented in several digital formats at different bit-rates, depending on the communications links over which the speech is transported.

A set of BTSs is controlled by a Base Station Controller (BSC). The BSC sets up and terminates calls to and from the BTSs and hands over ongoing calls among the BTSs based on the quality of the wireless links between the MS and the BTSs or the level of cell loading. The Mobile Switching Center (MSC) manages the operation of the controllers and connects them with either the Public Switched Telephone Network (PSTN) or other circuit-switched mobile communications networks. The 64 kbps Pulse Coded Modulation (PCM) format is typically used from the BSC and upward, i.e., in the direction of the MSC. The speech delivered to the MS undergoes the reverse signal processing operations.

The link between the MS and the BTS is not the only wireless link in the end-to-end speech transmission paths. In addition, microwave links, consisting of one or more T1 (1.544 Mbps) or E1 (2.024 Mbps) lines modulated in the high-frequency carriers, are often used as *backhaul* between the base station and the switching center, or in locations where fixed networks are not available or economical. Since the microwave is operated as a high-powered, line-of-sight wireless link over a dedicated, low-cost frequency spectrum, it does not suffer from many of the limitations inherent in mobile communications. In this chapter, we confine our interest to the dynamic interactions of the MS, BTS, BSC, and MSC, which must be carefully coordinated to maximize both speech quality and network capacity. Similar approaches will be taken in the following chapters but the network architectures or the node names will change as the mobile communications systems evolve.

The term *circuit-switched* refers to the nature of communications links in which the information, such as digitally formatted speech, is transmitted with negligible variation in speed or delay, regardless of the link quality or network load. This definition does not necessarily imply that the level of data loss or the bit-rate is uniformly maintained over the end-to-end transmission paths, however. A circuit-switched network consists of a series of such communications links, each of which transports the speech or data of one or more users at a fixed bit-rate. The end-to-end paths meeting the required transport capabilities and and channel conditions must be established before the transmission begins.

The interface between two communications links where the bit-rate or the speech format needs to be changed may require an additional processing delay but such a delay is generally lower than that associated with *packet-switched* networks such as the Internet or Ethernet, where the data packets can be transmitted without establishing an end-to-end transmission path. Without an established path, data packets can be lost or delivered in an order that is different from the order in which they were initially transmitted. The maximum allowed total delay, i.e., the mouth-to-ear delay, in commercial voice telephony systems is required to be equal to or less than 280 ms for a satisfactory call quality [ITU-T (2003)]. The wired telephone networks and contemporary circuit-switched mobile communications networks often complete the entire procedures in less than 200 ms.

In circuit-switched networks, the coded received speech first encounters error correction decoding, which is followed by error concealment when uncorrected errors corrupt the speech. The decoded speech is then converted to an analog representation for play out. Re-transmission of missing or corrupted frames, which would increase the delay and its variability, is generally not used. In packet-switched networks, each network node is allowed to retransmit lost data packets reported by a neighboring node, provided that the total delay budget is met. As interim solutions that bridged the gap between these two fundamentally different transmission techniques, hybrid approaches that combined the benefits of circuit-switched wired networks and packet-switched wireless networks were proposed and standardized [Ozturk *et al.* (2010)]. With these approaches, speech handling in the wired portions of the network is identical to that in conventional circuit-switched networks while re-transmission of lost speech data and scheduling of shared channels are allowed in the wireless links between the MS and the BTS.

Figure 1.2 shows the signal processing operations employed when the speech is transmitted between two second generation digital circuit-switched networks, from GSM to cdma2000. The digitized and compressed speech is wirelessly transmitted by the MS and recovered from the Radio Frequency (RF) signal by the BTS. The compressed speech is then reconstructed at the Transcoder and Rate Adaptation Unit (TRAU), which can be located at either the BTS, BSC, or MSC. The farther the TRAU is separated from the BTS, the farther the speech is transported at its lowest bit-rate. This saves the infrastructure cost since a 64 kbps channel can transport four speech channels encoded at bit-rates lower than 16 kbps. Therefore it is advantageous to extend the distance between the speech encoder and the speech decoder as far as possible, in some cases covering the entire transmission path. Voice over IP (VoIP) is an example of such an extreme case.

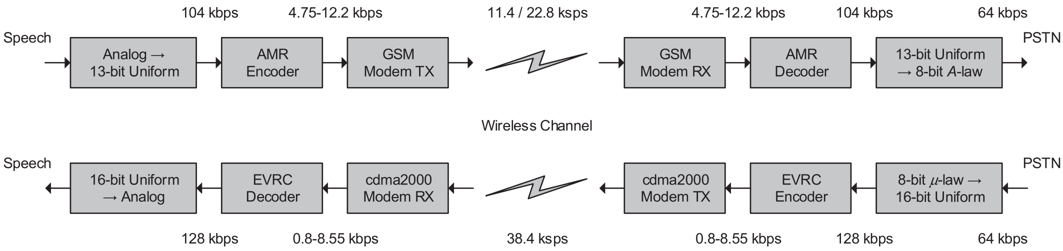

Fig. 1.2 Speech and radio signal processing operations from GSM to cdma2000.

The wireless link between the MS and the BTS is unique in that the bit-rate of the speech, which can change even during a call depending on the voice activity or the network control, is the lowest in the transmission path. Furthermore, because of the harsh nature of the wireless channel and the limited signal processing and transmit power of the MS, the speech is more likely to be damaged or lost in this short link than in any other. The transmission cost is also the highest in this link, because of the large investment for radio spectrum and network infrastructure.

The roaming capability, when extended globally, greatly increases the value of the radio spectrum that is shared by many countries. As a result, each generation of mobile communications systems has made more efficient use of the radio spectrum than its predecessors while simultaneously improving the speech quality. The main objectives of the signal processing operations in circuit-switched mobile communications networks can be simply summarized as the maximization of the number of satisfied users through efficient design of the network architectures and the procedures for all of the entities between the MS and the MSC. Beyond this point the existing PSTN infrastructure allows few opportunities for innovation.

Figure 1.3 illustrates the generic signal processing operations that occur between the MS and the BTS that are applicable to most digital circuit-switched networks. To counter the negative effects of the wireless channel including propagation loss and multipath fading of the transmitted signal, the MS and the BTS continuously control the bit-rate and transmit power, and report the channel status to the BSC so that the call can be transferred to a neighboring BTS with better link quality when the current BTS cannot support the necessary network services. The BTS, to which the call is handed over, may belong to the same or a different network type. During the transfer process, some of the speech signals en route to the destination BTS or MS can be lost, generating a small but audible loss of quality.

A number of metrics and criteria have been established to measure how well these performance objectives are met. This and the following chapters will show that these objectives can be achieved using a variety of approaches. These range from efficient speech compression algorithms that result in speech quality that is high enough for commercial services at low bit-rates to wireless communications techniques that use less bandwidth and/or less transmit power. In a restricted medium, such as the wireless channel, higher signal quality and higher network capacity are conflicting objectives. Thus, control mechanisms that trade one against the other play a key role in the overall system operation. New techniques for speech compression or wireless transmission need

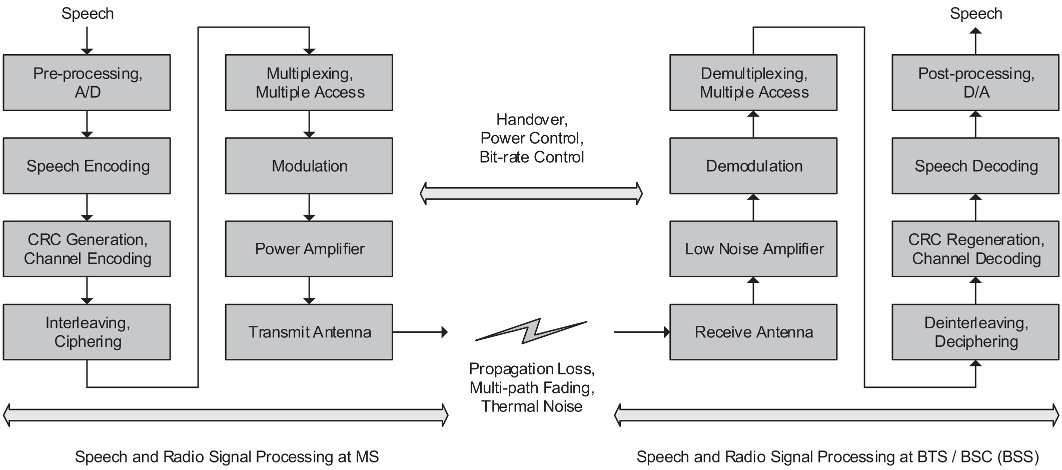

Fig. 1.3 Generic speech and radio signal processing operations in mobile communications systems.

to be incorporated carefully, however, to be compatible with the existing infrastructure. Changes can be made to operational networks but for MSs, once manufactured and activated, it can be very difficult, if not impossible, to make substantial changes other than software upgrades.

1.1.2 Performance Criteria

The speech and radio signal processing procedures used in mobile communications systems are designed to meet well-established criteria for maintaining speech quality. These fall into five types. The *blocked call rate* measures the capability of the network to handle incoming service requests. A request for setting up a call might be rejected because of insufficient radio resource or poor link quality. The blocked call rate does not differentiate among the possible sources of call blocking. This measure can be applied to a diversity of network types including fixed or mobile, analog or digital, and circuit-switched or packet-switched.

A second quality criterion is the *call drop rate*, which evaluates the capability of the network to maintain an established call. Conventional telephony systems such as the PSTN maintain a negligible call drop rate, but mobile communications systems are likely to exhibit rates as high as a few percent, regardless of the underlying radio access technologies or speech compression algorithms.

A third group of factors that affect the speech quality includes those that measure the reliability or link quality of the connection. These include the *bit error rate*, *frame error rate*, or *frame erasure rate*, all of which measure the probability that encoded speech frames are corrupted or lost in the channel during transmission. In the PSTN, the bit error rate is typically as low as 10^{-6} whereas a 1–3% corruption rate for transmitted speech frames is considered acceptable in mobile communications networks. When the received speech frames contain bit errors, the error control coding may identify the location of those errors and recover the speech. Error concealment methods, such as the

Cambridge University Press
978-1-108-42103-4 — Media and Radio Signal Processing for Mobile Communications
Kyunghun Jung , Russell M. Mersereau
Excerpt
More Information

methods that replace corrupted frames by interpolating or extrapolating nearby correctly decoded speech frames, can also be used to maintain an acceptable speech quality.

A fourth group of quality criteria relates to the operation of packet-switched networks, especially those built with the Internet Protocol (IP). This group includes such measures as the *packet loss rate* and *jitter loss rate* to evaluate the effects of different error types. Finally, there are measures of network acceptability that quantify the *end-to-end delay* of the voice services. These are often the most stringent to meet but they have a profound influence on the overall design of the system.

These five groups of quality criteria are defined mainly to establish a set of minimum requirements for *toll quality* or *carrier grade* services. In many cases these are objectively measurable but they cannot completely replace the important subjective criteria, as measured by the Mean Opinion Score (MOS) derived from subjective evaluations with human listeners. From the point of view of service providers, all of these criteria are used to maximize the number of simultaneous calls whose quality exceeds a set of minimum requirements, rather than to maximize the quality of each call for a fixed number of simultaneous calls.

1.2 Analog Mobile Communications Systems

When it was developed in the 1970s and commercially launched in 1982, the Advanced Mobile Phone System (AMPS) introduced many fundamental aspects of mobile communications systems, such as the frequency re-use to increase network capacity and the handover of ongoing calls between cells [Young (1979), MacDonald (1979)]. Some aspects were necessary to cope with the regulatory limitations. One of these was the restricted bandwidth that was allocated. The AMPS was initially assigned two 25 MHz bands located above 800 MHz for the forward (BTS to MS), and reverse (MS to BTS) channels. The AMPS was adopted by many countries and often operated in frequency bands slightly different from the original ones.

For the Frequency Modulation (FM) techniques that were used in AMPS, radio spectrum below 800 MHz would have been preferred but was not then available. The 800 MHz bands were from a part of the radio spectrum that had previously been occupied by television channels. This bandwidth had been freed after the channels were relocated to cable. When the number of people using mobile communications continued to increase, the need to accommodate additional customers, coupled with the difficulty of obtaining additional spectrum, resulted in technical decisions made during the redesign of AMPS that influenced many key aspects of the next generation digital mobile communications systems.

Before proceeding to more detailed descriptions of AMPS, it is important to distinguish between a *band* and a *channel* as these terms are used in this book. We follow the definitions of [Razavi (2011)], in which a band refers to the entire radio spectrum in which the MSs of a mobile communications system are allowed to communicate, while a channel refers to the smaller bandwidth assigned to one or more MSs for services. These definitions match well with the spectrum allocation practices of both the

Table 1.1 Channel numbering system.

Channel number n	Reverse channel frequency (MHz)	Forward channel frequency (MHz)
1–799	$825 + 0.03n$	$870 + 0.03n$
991–1023	$825 + 0.03(n - 1023)$	$870 + 0.03(n - 1023)$

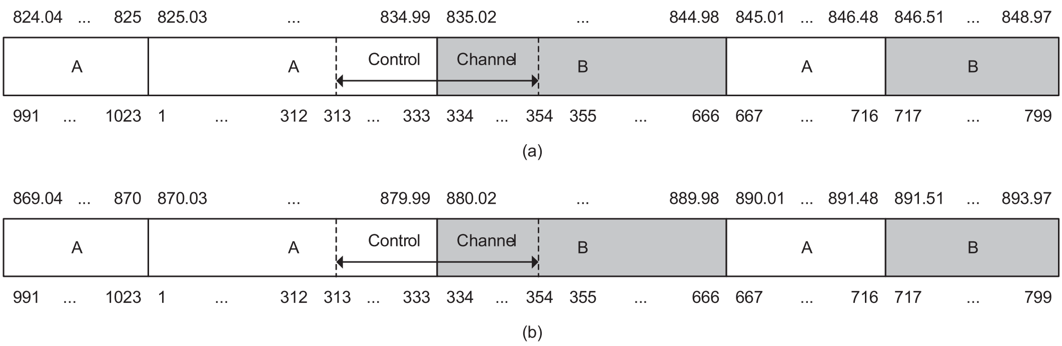

Fig. 1.4 Spectrum allocation. (a) Reverse channels. (b) Forward channels.

first generation analog mobile communications systems and the second generation Time Division Multiple Access (TDMA) systems. In these systems, signals from one or more MSs are transmitted over a narrowly confined channel of 30–200 kHz. Each channel has the center frequency of an RF carrier and an integer is typically assigned to label each channel. Then, a set of contiguous channels constitutes a band. An MS in a mobile communications system requires at least one band for the reverse channels and another for the forward channels, if it is operated in the Frequency Division Duplex (FDD) mode.

1.2.1 Network Architecture

With a 25 MHz band and a channel spacing of 30 kHz, AMPS provides 832 channels that can be divided between one or more service providers in each area. A typical configuration might be that a half of the total capacity, i.e., a combination of 395 channels for voice service and 21 channels for call control, would be assigned to each service provider in a market where two providers compete. Figure 1.4 shows the spectrum allocation of AMPS in the US in the 1980s when two types of service providers, a non-wireline (A) operator and a wireline (B) operator, shared the band. Channels 313–333 and 334–354 are the control channels assigned for each operator. The channel numbers and carrier frequencies of AMPS are related as shown in Table 1.1.

The earlier architecture of a generic circuit-switched mobile communications network, shown in Fig. 1.1, is directly applicable to AMPS whose network architecture is shown in Fig. 1.5; only the terminology is different. In AMPS, the Base Station (BS) performs similar network operations as the BTS, and the Mobile Telephone Switching Office (MTSO) performs management tasks similar to those of the BSC and the MSC. It controls the call processing and manages the cellular operation.

Cambridge University Press
978-1-108-42103-4 — Media and Radio Signal Processing for Mobile Communications
Kyunghun Jung , Russell M. Mersereau
Excerpt
More Information

Table 1.2 Control partitioning of AMPS.

MS	BS	MTSO
Setup channel selection Channel tuning Message reception and transmission Failure sequence timing Tone switch-hook supervision Pre-origination dialing	Radio control Location data collection Component calibration MS control Message relaying and reformatting Switch-hook and fade supervision	Standard local switching Radio channel management Remote unit maintenance BS and MS control Message administration MS location tracking Handover synchronization

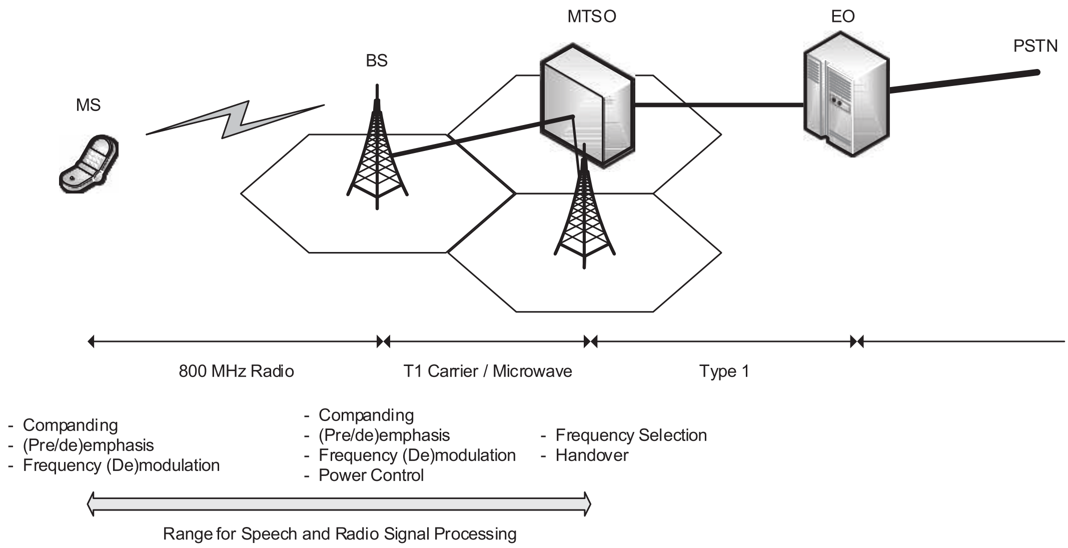

Fig. 1.5 Network architecture of AMPS.

MTSO can be interconnected to a Local Exchange Carrier (LEC) End Office (EO) with Type 1 interconnection link. Table 1.2 outlines the technical responsibilities of the MS, BS, and MTSO partitioned among these three entities [Fluhr and Porter (1979)]. Although AMPS is often classified as an analog mobile communications system, many signal processing and control channel operations are represented in digital formats.

Figure 1.5 also shows the interface types used between the network entities of AMPS. We focus on the link between the MS and the BS, and the link between the BS and the MTSO. The first link is of crucial importance since no further signal processing is performed after the analog speech waveform from the MS is digitized and encoded into a 64 kbps PCM format at the BS. This basic format for speech is maintained throughout the transmission paths until the signal reaches either another BS or a fixed telephone.

Conversion between two PCM formats, μ-law and A-law, may occur at intermediate locations but this would have little impact on the speech quality or total delay since the two formats are similarly defined and the conversion requires a negligible amount of computation. A T1 carrier, microwave, or Type 1 interconnection link carries large numbers of 64 kbps PCM channels. The second link, between the BS and the MTSO, is also important since the MTSO is responsible for controlling speech quality and network capacity by indirectly controlling the MS through the BS. The measures available to the MTSO include the handover to another BS or channel in the same cell, and power control.

1.2.2 Speech and Radio Signal Processing Operations

Figure 1.6 shows the speech and radio signal processing operations in AMPS for the transmit and receive sides [Arredondo *et al.* (1979)]. In the first step, the sound pressure level of speech is converted to voltage variations by the microphone, and then band-pass filtering limits the bandwidth of the signal to 300–3000 Hz. Because the waveform will be frequency modulated, the signal amplitude is also limited to control the amount of energy that would be leaked to adjacent channels. This is done by companding, i.e., non-linearly compressing the amplitude at the transmitter and expanding it at the receiver. AMPS uses a 2:1 compander, through which a 2 dB change in the input voltage level is compressed to a 1 dB change. The compander also has the effect of improving the subjective speech quality in poor channel conditions. Figure 1.7 illustrates the companding and modulation procedures used.

The energy of speech signal, after filtering and compression, is concentrated in the low frequency bands and the Signal-to-Noise Ratio (SNR) in the high frequency bands is reduced. It is further degraded in the FM and carrier modulation process. Figures 1.7(a) and 1.7(b) show the input-output characteristics of the compander and the deviation limiter, respectively. With a channel width of 30 kHz, the frequency deviation of speech signal is confined to approximately 24 kHz, around a center frequency f_c,

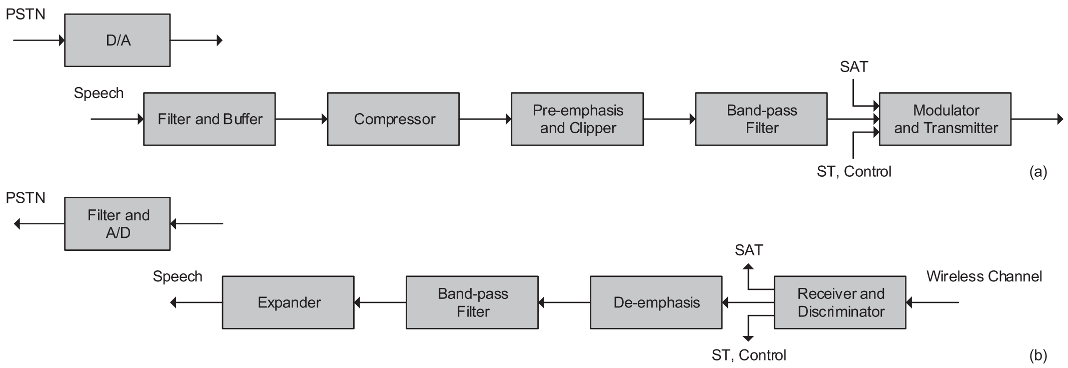

Fig. 1.6 Speech and radio signal processing operations of AMPS. (a) Transmitter side. (b) Receiver side.

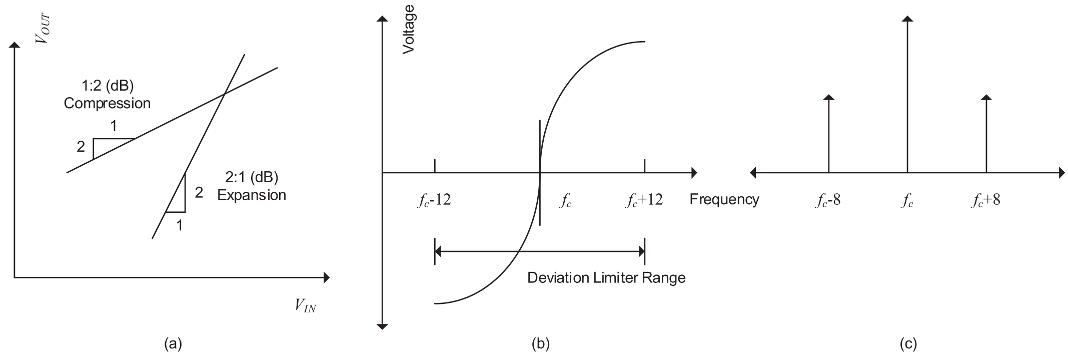

Fig. 1.7 (a) Compander input-output characteristics. (b) Frequency deviation limiting. (c) ±8 kHz binary FSK.

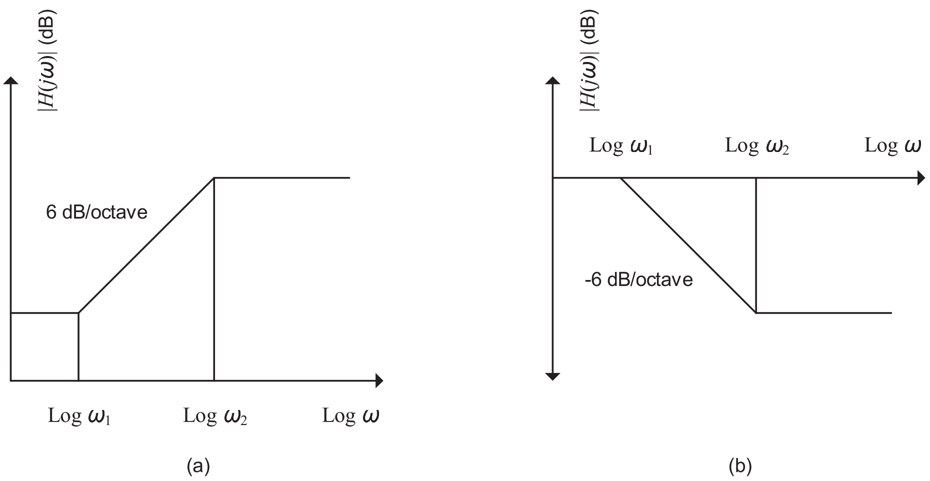

Fig. 1.8 Frequency response. (a) Pre-emphasis. (b) De-emphasis.

to reduce the interference to and from the adjacent channels. Pre-emphasis boosts the high-frequency components of speech signal at the transmitter while de-emphasis compensates for this at the receiver. Figures 1.8(a) and 1.8(b) show the frequency response of the pre-emphasis and de-emphasis filters, respectively, where the angular frequency $\omega = 2\pi f$. These analog signal processing operations are common to both the forward and reverse channels of AMPS.

There are no measures that prevent unauthorized eavesdropping of ongoing calls in AMPS, since providing reliable security with analog signal processing is very difficult. Anyone with the intention and capability of scanning channels can listen to or record the conversations. This fundamental limitation of AMPS was overcome in the next generation digital systems in which ciphering of digitally compressed speech became a basic feature.