CELLULAR & MOBILE COMMUNICATION

LECTURE NOTES

B.TECH (IV-I SEM)

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UNIT-V

HANDOFFS

WHY HAND OFF IS NECESSARY

In an analog system, once a call is established, the set-up channel is not used again during the call period. Therefore, handoff is always implemented on the voice channel. In the digital systems, the handoff is carried out through paging or common control channel. The value of implementing handoffs is dependent on the size of the cell. For example, if the radius of the cell is 32 km (20 mi), the area is 3217 km^2(1256 mi^2). After a call is initiated in this area, there is little chance that it will be dropped before the call is terminated as a result of a weak signal at the coverage boundary. Then why bother to implement the handoff feature? Even for a 16-km radius, cell handoff may not be needed. If a call is dropped in a fringe area, the customer simply redials and reconnects the call. Today the size of cells becomes smaller in order to increase capacity. Also people talk longer. The handoffs are very essential. Handoff is needed in two situations where the cell site receives weak signals from the mobile unit: (1) at the cell boundary, say, −100 dBm, which is the level for requesting a handoff in a noise-limited environment; and (2) when the mobile unit is reaching the signal-strength holes (gaps) within the cell site as shown in Fig.1.

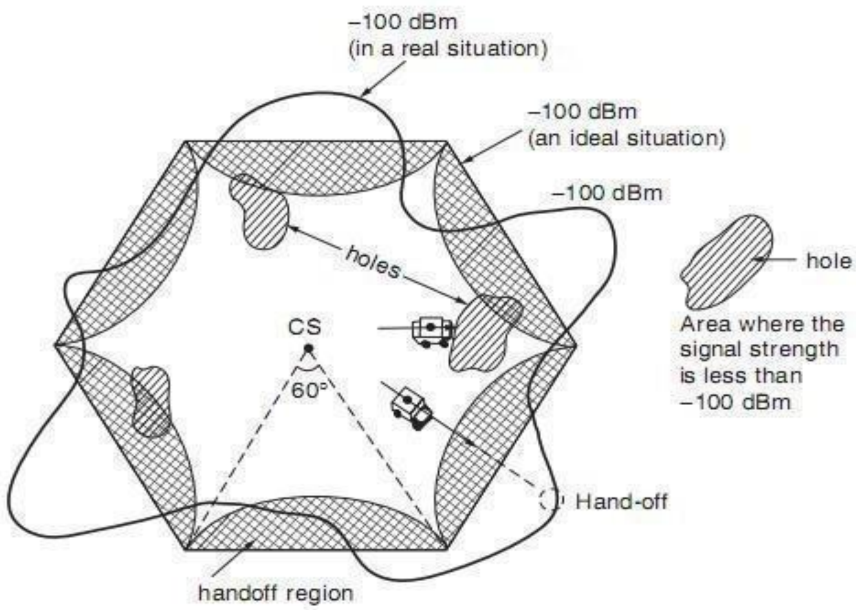


Fig.1. Occurrence of handoffs

WHAT ARE THE TWO DECISIONS MAKING PARAMETERS OF HANDOFF EXPLAIN

There are two decision-making parameters of handoff: (1) that based on signal strength and (2) that based on carrier-to-interference ratio. The handoff criteria are different for these two types. In type 1, the

signal-strength threshold level forhandoffis −100dBminnoise-limited systems and −95dBm ininterference-limited systems. In type 2, the value of C/I at the cell boundary for handoff should be at a level, 18 dB for AMPS
in order to have toll quality voice. Sometimes, a low value of C/I may be used for capacity reasons.

Type 1: It is easy to implement. The location receiver at each cell site measures all the signal strengths of all receivers at the cell site. However, the received signal strength (RSS) itself includes interference.

$$RSS = C + I$$

where C is the carrier signal power and I is the interference. Suppose that we set up a threshold level for RSS; then, because of the I , which is sometimes very strong, the RSS level is higher and far above the handoff threshold level. In this situation handoff should theoretically take place but does not. Another situation is when I is very low but RSS is also low. In this situation, the voice quality usually is good even though the RSS level is low, but since RSS is low, unnecessary handoff takes place. Therefore, it is an easy but not very accurate method of determining handoffs. Some analog systems use SAT information together with the received signal level to determine handoffs. Some CDMA systems use pilot channel information.

Type 2: Handoffs can be controlled by using the carrier-to-interference ratio C/I C+I/I = C/I

we can set a level based on C/I ,so C drops as a function of distance but I is dependent on the location. If the handoff is dependent on C/I , and if the C/I drops, it does so in response to increase in (1) propagation distance or
(2) interference. In both cases, handoff should take place. In today's cellular systems, it is hard to measure C/I during a call because of analog modulation. Sometimes we measure the level I before the call is connected, and the level C + I during the call. Thus (C + I)/I can be obtained.

TYPES OF HANDOFF

There are four types of handoff:

1. **INTERSECTOR OR SOFTER HANDOFF.**

   The mobile communicates with two sectors of the same cell (see Fig. 10-1). A RAKE receiver at the base station combines the best versions of the voice frame from the diversity antennas of the two sectors into a single traffic frame.

2. **INTERCELL OR SOFT HANDOFF.**

   The mobile communicates with two or three sectors of different cells (see Fig. 10-2). The base station that has the direct control of call processing during handoff is referred to as the primary base station. The primary base station can initiate the forward control message. Other base stations that do not have control over call processing are called the secondary base stations. Soft handoff ends when either the primary or secondary base station is dropped. If the primary base station is dropped, the secondary base station becomes the new primary for this call. A three-way soft handoff may end by first dropping one of the base stations and becoming a two-way soft handoff. The base stations involved coordinate handoff by exchanging information via SS7 links. A soft handoff uses considerably more network resources than the softer handoff.

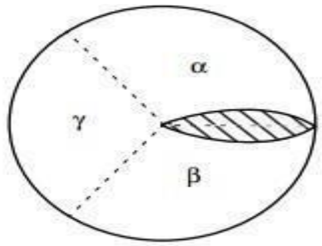


Figure 10-1 Softer Handoff

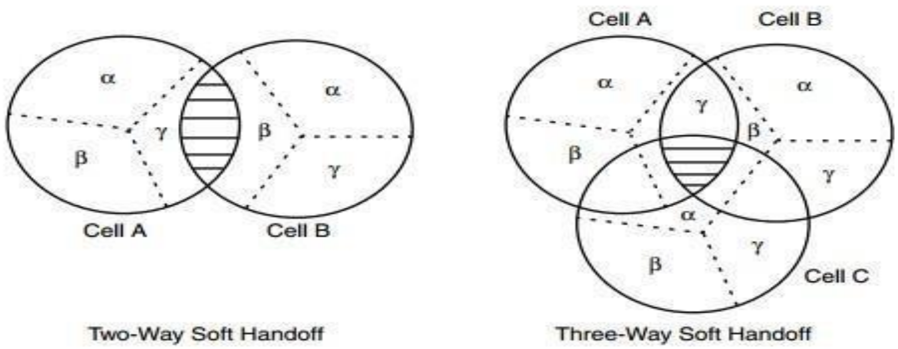


Figure 10-2 Soft Handoff

3. SOFT-SOFTER HANDOFF.

The mobile communicates with two sectors of one cell and one sector of another cell (see Fig. 10-3). Network resources required for this type of handoff include the resources for a two-way soft handoff between cell A and B plus the resources for a softer handoff at cell B.

4. HARD HANDOFF.

Hard handoffs are characterized by the break-before-make strategy. The connection with the old traffic channel is broken before the connection with the new traffic channel is established. Scenarios for hard handoff include

◆ Handoff between base stations or sectors with different CDMA carriers

◆ Change from one pilot to another pilot without first being in soft handoff with the new pilot (disjoint active sets)

◆ Handoff from CDMA to analog, and analog to CDMA

◆ Change of frame offset assignment—CDMA traffic frames are 20 ms long. The start of frames in a particular traffic channel can be at 0 time in reference to a system or it can be offset by up to 20 ms (allowed in IS- 95). This is known as the frame offset. CDMA traffic channels are assigned different frame offset to avoid congestion. The frame offset for a particular traffic channel is communicated to the mobile. Both forward and reverse links use this offset. A change in offset

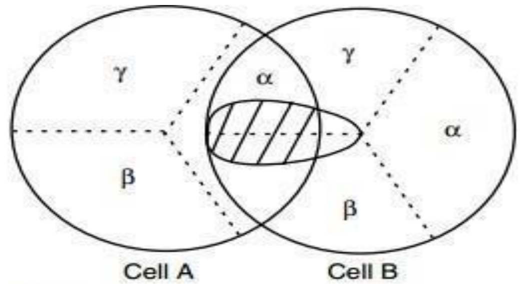


Figure 10-3 Soft-Softer Handoff

Assignment will disrupt the link. During soft handoff the new base station must allocate the same frame offset to the mobile as assigned by the primary base station. If that particular frame offset is not available, a hard handoff may be required. Frame offset is a network resource and can be used up

HANDOFF INITIATION

A hard handoff occurs when the old connection is broken before a new connection is activated. The performance evaluation of a hard handoff is based on various initiation criteria [1, 3, 13]. It is assumed that the signal is averaged over time, so that rapid fluctuations due to the multipath nature of the radio environment can be eliminated. Numerous studies have been done to determine the shape as well as the length of the averaging window and the older measurements may be unreliable. Figure 1.2 shows a MS moving from one BS (BS1) to another (BS2). The mean signal strength of BS1 decreases as the MS moves away from it. Similarly, the mean signal strength of BS2 increases as the MS approaches it. This figure is used to explain various approaches described in the following subsection.

RELATIVE SIGNAL STRENGTH

This method selects the strongest received BS at all times. The decision is based on a mean measurement of the received signal. In Figure 1.2, the handoff would occur at position A. This method is observed to provoke too many unnecessary handoffs, even when the signal of the current BS is still at an acceptable level.

RELATIVE SIGNAL STRENGTH WITHTHRESHOLD

This method allows a MS to hand off only if the current signal is sufficiently weak (less than threshold) and the other is the stronger of the two. The effect of the threshold depends on its relative value as compared to the signal strengths of the two BSs at the point at which they are equal. If the threshold is higher than this value, say T1 in Figure 1.2, this scheme performs exactly like the relative signal strength scheme, so the handoff occurs at position

A. If the threshold is lower than this value, say T2 in Figure 1.2, the MS would delay handoff until the current signal level crosses the threshold at position B. In the case of T3, the delay may be so long that the MS drifts too far into the new cell.

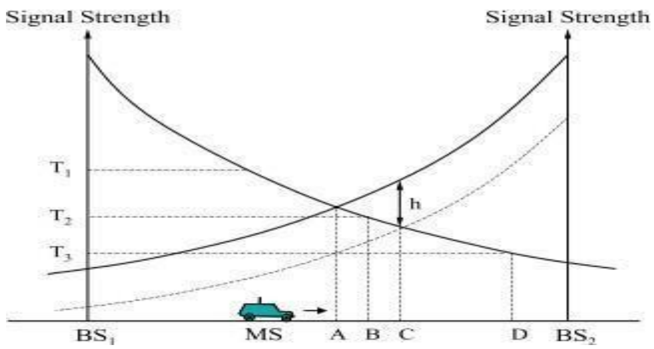


Figure 1.2 Signal strength and hysteresis between two adjacent BSs for potential handoff.

This reduces the quality of the communication link from BS1 and may result in a dropped call. In addition, this results in additional interference to cochannel users. Thus, this scheme may create overlapping cell coverage areas. A threshold is not used alone in actual practice because its effectiveness depends on prior knowledge of the crossover signal strength between the current and candidate BSs.

RELATIVE SIGNAL STRENGTH WITH HYSTERESIS

This scheme allows a user to hand off only if the new BS is sufficiently stronger (by a hysteresis margin, h in Figure 1.2) than the current one. In this case, the handoff would occur at point C. This technique prevents the so- called ping-pong effect, the repeated handoff between two BSs caused by rapid fluctuations in the received signal strengths from both BSs. The first handoff, however, may be unnecessary if the serving BS is sufficiently strong.

RELATIVE SIGNAL STRENGTH WITH HYSTERESIS AND THRESHOLD

This scheme hands a MS over to a new BS only if the current signal level drops below a threshold and the target BS is stronger than the current one by a given hysteresis margin. In Figure 1.2, the handoff would occur at point D if the threshold is T3

PREDICTION TECHNIQUES

Prediction techniques base the handoff decision on the expected future value of the received signal strength. A technique has been proposed and simulated to indicate better results, in terms of reduction in the number of unnecessary handoffs, than the relative signal strength, both without and with hysteresis, and threshold methods.

CONCEPT OF DELAYING A HANDOFF

In many cases, a two-handoff-level algorithm is used. The purpose of creating two request handoff levels is to provide more opportunity for a successful handoff. A handoff could be delayed if no available cell could take the call. A plot of signal strength with two request handoff levels and a threshold level is shown in Fig.3. The plot of average signal strength is recorded on the channel received

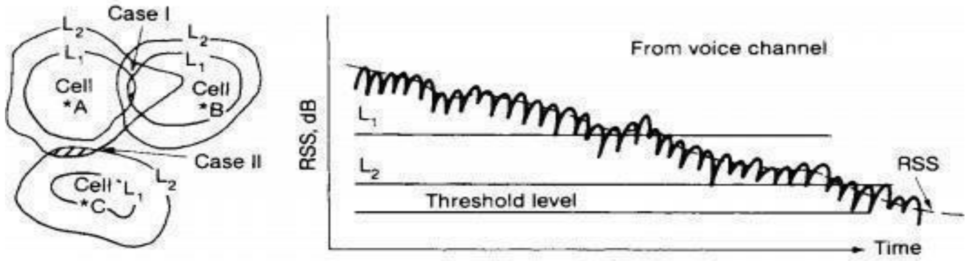


Fig.3. A two level handoff scheme

Signal strength indicator (RSSI), which is installed at each channel receiver at the cell site. When the signal strength drops below the first handoff level, a handoff request is initiated. If for some reason the mobile unit is in a hole (a weak spot in a cell) or a neighboring cell is busy, the handoff will be requested periodically every 5 s. At the first handoff level, the handoff takes place if the new signal is stronger. However, when the second handoff level is reached, the call will be handed off with no condition. The MSO always handles the handoff call first and the originating calls second. If no neighboring calls are available after the second handoff level is reached, the call continues until the signal strength drops below the threshold level; then the call is dropped. In AMPS systems if the supervisory audio tone (SAT) is not sent back to the cell site by the mobile unit within 5 s, the cell site turns off the transmitter.

ADVANTAGES OF DELAYED HANDOFF

1. Consider the following example. The mobile units are moving randomly and the terrain contour is uneven. The received signal strength at the mobile unit fluctuates up and down. If the mobile unit is in a hole for less than 5 s (a driven distance of 140 m for 5 s, assuming a vehicle speed of 100 km/h), the delay (in handoff) can even circumvent the need for a handoff. If the neighboring cells are busy, delayed handoff may take place. In principle, when call traffic is heavy, the switching processor is loaded, and thus a lower number of handoffs would help the processor handle call processing more adequately. Of course, it is very likely that after the second handoff level is reached, the call may be dropped with great probability.
2. The other advantage of having a two-handoff-level algorithm is that it makes thehandoff occur at the proper location and eliminates possible interference in the system. Figure 3, case I, shows the area where the first-level handoff occurs between cell A and cell B. If we only use the second-level handoff boundary of cell A, the area of handoff is too close to cell B. Figure 3, case II, also shows where the second-level handoff occurs between cell A and cell C. This is because the first-level handoff cannot be implemented.

POWER DIFFERENCE HANDOFF

A better algorithm is based on the power difference (_) of a mobile signal received by two cell sites, home and handoff. _ can be positive or negative. The handoff occurs depending on a preset value of _.

_ = the mobile signal measured at the candidate handoff site

− The mobile signal measured at the home

site For example, the following cases can occur.

_> 3 dB request a handoff

1dB <_< 3 dB prepare a handoff

−3dB <_< 0 dB monitoring the signal strength

_< −3 dB no handoff

Those numbers can be changed to fit the switch processor capacity. This algorithm is not based on the received signal strength level, but on a relative (power difference) measurement. Therefore, when this algorithm is used, all the call handoffs for different vehicles can occur at the same general location in spite of different mobile antenna gains or heights.

FORCED HANDOFF

A forced handoff is defined as a handoff that would normally occur but is prevented from happening, or a handoff that should not occur but is forced to happen.

MOBILE-ASSISTED HANDOFF

In a mobile-assisted handoff process, the MS makes measurements and the network makes thedecision. In the circuit switched GSM (global system mobile), the BS controller (BSC) is in charge of the radio interface management. This mainly means allocation and release of radio channels and handoff management. The handoff time between handoff decision and execution in such a circuit-switched GSM is approximately 1 second.

SOFT HANDOFF

SOFT HANDOFF (FORWARD LINK)

In this case all traffic channels assigned to the mobile are associated with pilots in the active set and carry the same traffic information with the exception of power control subchannel. When the active set contains more than one pilot, the mobile provides diversity by combining its associated forward traffic channels.

SOFT HANDOFF (REVERSE LINK)

During intercell handoff, the mobile sends the same information to both base stations. Each base station receives the signal from the mobile with appropriate propagation delay. Each base station then transmits the received signal to the vocoder/selector. In other words, two copies of the same frame are sent to the vocoder/selector. The vocoder/selector selects the better frame and discards the other.

SOFTER HANDOFF (REVERSE LINK)

During intersector handoff, the mobile sends the same information to both sectors. The channel card/element at the cell site receives the signals from both sectors. The channel card combines both inputs, and only one frame is sent to the vocoder/selector. It should be noted that extra channel cards are not required to support softer handoff as is the case for soft handoffs. The diversity gain from soft handoffs is more than the diversity gain from softer handoffs because signals from distinct cells are less correlated than signals from sectors of the same cell.

10.2.4 BENEFIT OF SOFT HANDOFF

A key benefit of soft handoff is the path diversity on the forward and reverse traffic channels. Diversity gain is obtained because less power is required on the forward and reverse links. This implies that total system interference is reduced. As a result, the average system capacity is improved. Also less transmit power from the mobile results in longer battery life and longer talk time. In a soft handoff, if a mobile receives an up power control bit from one base station and a down control bit from the second base station, the mobile decreases its transmit power. The mobile obeys the power down command since a good communications link must have existed to warrant the command from the second base station.

INTERSYSTEM HANDOFF

Occasionally, a call may be initiated in one cellular system (controlled by one MSO) and enter another system (controlled by another MSO) before terminating. In some instances, intersystem handoff can take place; this means that a call handoff can be transferred from one system to a second system so that the call is continued while the mobile unit enters the second system. The software in the MSO must be modified to apply this situation. Consider the simple diagram shown in Fig.7. The car travels on a highway and the driver originates a call in system A. Then the car leaves cell site A of system A and enters cell site B of system B. Cell sites A and B are controlled by two different MSOs. When the mobile unit signal becomes weak in cell site A, MSO A searches for a candidate cell site in its system and cannot find one. Then MSO A sends

The handoff request to MSO B through a dedicated line between MSO A and MSO B, and MSO B makes a complete handoff during the call conversation. This is just a one-point connection case. There are many ways of implementing intersystem handoffs, depending on the actual circumstances. For instance, if two MSOs are manufactured by different companies, then compatibility must be determined before implementation of intersystem handoff
can be

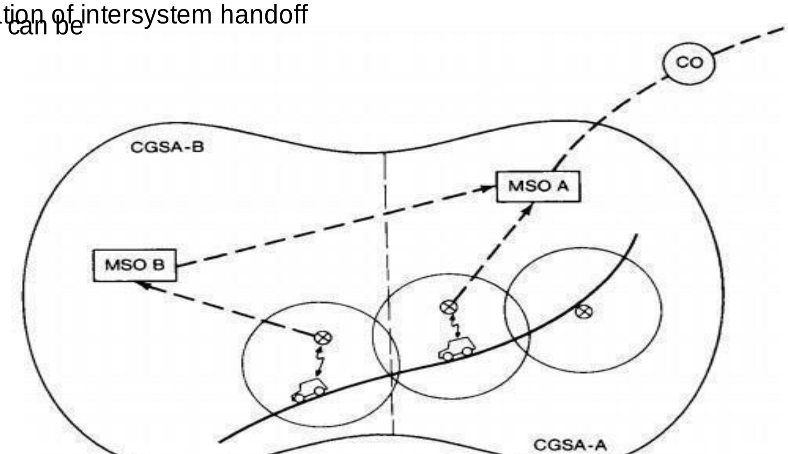


Fig.7. Intersystem handoffs

DEFINITION OF DROPPED CALL RATE AND CONSIDERATION OF DROPPED CALL RATES

The definition of a dropped call is after the call is established but before it is properly terminated. The definition of "the call is established" means that the call is setup completely by the setup channel. If there is a possibility of a call drop due to no available voice channels, this is counted as a blocked call not a dropped call. If there is a possibility that a call will drop due to the poor signal of the assigned voice channel, this is considered a dropped call. This case can happen when the mobile or portable units are at a standstill and the radio carrier is changed from a strong setup channel to a weak voice channel due to the selective frequency fading phenomenon.

The perception of dropped call rate by the subscribers can be higher due to:

1. The subscriber unit not functioning properly (needs repair).

2. The user operating the portable unit in a vehicle (misused).

3. The user not knowing how to get the best reception from a portable unit (needs education).

In principle, dropped call rate can be set very low if we do not need to maintain the voice quality. The dropped call rate and the specified voice quality level are inversely proportional. In designing a commercial system, the specified voice quality level is given relating to how much C/I (or C/N) the speech coder can tolerate. By maintaining a certain voice quality level, the dropped call rate can be calculated by taking the following factors into consideration:

1. Provide signal coverage based on the percentage (say 90 percent) that the entire received signal will be above a given signal level.

2. Maintain the specified co-channel and adjacent channel interference levels in each cell during a busy hour (i.e., the worst interference case).

3. Because the performance of the call dropped rate is calculated as possible call dropping in every stage from the radio link to the PSTN connection, the response time of the handoff in the network will be a factor when the cell becomes small, the response time for a handoff request has to be shorter in order to reduce the call dropped rate.

RELATION AMONG CAPACITY, VOICE QUALITY, DROPPED CALL RATE

Radio Capacity m is expressed as follows:

$$m = \frac{B_T / B_c}{\sqrt{\frac{2}{3}(C/I)_S}}$$

Where BT/Bc is the total number of voice channels. BT /Bc is a given number, and (C/I) s is a required C/I for designing a system. The above equation is obtained based on six co-channel interferers which occur in busy traffic (i.e., a worst case). In an interference limited system, the adjacent channel interference has only a secondary effect.

$$(C/I)_S = \frac{3}{2}\left(\frac{B_T/B_c}{m}\right)^2 = \frac{3}{2}\left(\frac{B_T}{B_c}\right)^2 \cdot \frac{1}{m^2}$$

Because the (C/I)s is a required C/I for designing a system, the voice quality is based on

the (C/I)s .When the specified (C/I)S is reduced, the radio capacity is increased. When the measured (C/I) is less than the specified (C/I)S , both poor voice quality and dropped calls can occur.

GENERAL FORMULA OF DROPPED CALL RATE

The general formula of dropped call rate P in a whole system can be expressed as:

$$P = 1 - \left[\sum_{n=0}^{N} \alpha_n X^n\right] = \sum_{n=0}^{N} \alpha_n \cdot P_n$$

Where

$$P_n = 1 - X^n$$

P_n is the probability of a dropped call when the call has gone through n handoffs and

$X = (1-\delta)(1-\mu)(1-\theta\tau)(1-\beta)^2$

δ = Probability that the signal is below the specified receive threshold (in a noise-limited system).

μ = Probability that the signal is below the specified cochannel interference level (in an interference-limited system).

τ = Probability that no traffic channel is available upon handoff attempt when moving into a new cell.

θ = Probability that the call will return to the original cell.

β = Probability of blocking circuits between BSC and MSC during handoff.

α_n = The weighted value for those calls having n handoffs, and $\sum_{n=0}^{N} \alpha_n = 1$

N = N is the highest number of handoffs for those calls.

1. z1 and z2 are two events, z1 is the case of no traffic channel in the cell, z2 is the case of no-safe return to original cell. Assuming that z1 and z2 are independent events, then

2. (1 − β) is the probability of a call successfully connecting from the old BSC to the MSC. Also, (1 − β) is the probability of a call successfully connecting from the MSC to

3. $P(z_2|z_1)\cdot P(z_1) = P(z_2)\cdot P(z_1) = \theta\cdot\tau$

4. the new BSC. Then the total probability of having a successful call connection is

$$\left.\begin{array}{ll} \text{BSC (old)} \rightarrow \text{MSC} & (1-\beta) \\ \text{MSC} \rightarrow \text{BSC (new)} & (1-\beta) \end{array}\right\} \rightarrow (1-\beta)^2$$

3. The call dropped rate P expressed in above Eq can be specified in two cases:

1. In a noise limited system (startup system): there is no frequency reuse, the call dropped rate PA is based on the signal coverage. It can also be calculated under busy hour conditions.

In a noise-limited environment (for worst case)

$$\delta = \delta_1$$

$$\mu = \mu_1$$

$$\left.\begin{aligned} \tau &= \tau_1 \\ \theta &= \theta_1 \\ \beta &= \beta_1 \end{aligned}\right\} \text{the conditions for the noise limited case}$$

2. In an interference-limited system (mature system): frequency reuse is applied, and the dropped rate PB is based on the interference level. It can be calculated under busy hour conditions.

In an interference-limited environment (for worst case)

$$\delta = \delta_2$$

$$\mu = \mu_1$$

$$\left.\begin{aligned} \tau &= \tau_2 \\ \theta &= \theta_2 \\ \beta &= \beta_2 \end{aligned}\right\} \text{the conditions for the interference limited case}$$

In a commonly used formula of dropped call rate, the values of τ , θ, and β are assumed to be very small and can be neglected. Then

$$X = (1 - \delta)(1 - \mu)$$

Furthermore, in a noise-limited case, $\mu \rightarrow 0$,

$$P_A = \sum_{n=0}^{N} \alpha_n P_n = \sum \alpha_n [1 - (1 - \delta)^n]$$

and in an interference-limited system, $\delta \rightarrow 0$,

$$P_B = \sum^{N} \alpha_n P_n = \sum \alpha_n [1 - (1 - \mu)^n]$$

13.1 Performance Evaluation

13.1.1 Blockage

There are two kinds of blockage: set-up channel blockage and voice-channel blockage.

Set-up channel blockage B_1. Information regarding set-up channel blockage cannot be obtained at the cell site because the mobile unit will be searching for the busy/idle bit of a forward set-up channel in order to set up its call. If the busy bit does not change after 10 call attempts in 1 s, a busy tone is generated, and no mobile transmit takes place. In another case the mobile transmit takes place as soon as the idle bit is shown. Several initiating cells can intercollide at the same time. When it occurs, the mobile unit counts it as one seizure attempt. If the number of seizure attempts exceeds 10, then the call is blocked. This kind of blockage can be detected only by mobile phone users. If the occurrence of blockage of the system is in doubt, each of the three specified set-up channels can be assigned in each of the three sectors of a cell, and the total number of incoming calls among the three sectors can be compared with that from a single set-up channel (omni). It should be determined whether there is a difference between two call-completion numbers, one from a single set-up channel and the other from three set-up channels. This is one way to check the blockage if the single set-up channel seems too busy. The set-up channel blockage should be at least less than half of the specified blockage (usually 0.02) in the mobile cellular system.

If all the call-attempt repeats are independent events, then the resultant blocking probability B_1 after n attempts is related to the blocking probability of the single call attempt B, as

$$B_1 = 1 - (1 - B) \sum_{i=0}^{n} B^i = B^n \qquad (13.1\text{-}1)$$

Example 13.1 Assume that the blocking probability of a set-up channel is .005, and the holding time at the set-up channel is 175 ms per call. There is only one channel; then the offered load (from Appendix 1.1) a is .005. Thus the number of set-up calls being handled is

$$C = \frac{.005 \times 3600 \times 1000}{175} = 120 \text{ calls} \qquad \text{(one call attempt)}$$

Voice-channel blockage B_2. Voice-channel blockage can be evaluated at the cell site. When all calls come in, some are refused for service because there are no available voice channels. Suppose that we are designing a voice channel blockage to be .02. On this basis, $B_2 = .02$, and after determining the holding time per call[1] and roughly estimating the total number of calls per hour at the site,[2] we can find the number of radios required.

Example 13.4 Assume that 2000 calls per hour are anticipated. The average holding time is 100 s per call, and the blocking probability is .02 (2 percent). Then the offered load is

$$a = \frac{2000 \times 100 \text{ s}}{60 \times 60 \text{ s}} = 55.5 \text{ erlangs}$$

Use $a = 55.5$ and $B_2 = 0.02$ to find $N = 66$ channels required (refer to Appendix 1.1).

The actual blocking probability data must be used to check the outcome from the Erlang B model (Appendix 1.1). Although the difference can be up to 15 percent, the Erlang B model is still considered as a good model for obtaining useful estimates.

End-office trunk blockage B_3. The trunks connecting from the MTSO to the end office can be blocked. This usually occurs when the call traffic starts to build up and the number of trunks connected to the end office becomes inadequate. Unless this corrective action is taken, the blockage during busy periods increases. An additional number of trunks could be provided at the end office when needed.

The total blockage B_t. As the total call blockage is the result of all three kinds of blockage, the total blockage is

$$\begin{aligned} B_t &= B_1 + B_2(1 - B_1) + B_3(1 - B_1)(1 - B_2) \\ &= 1 - (1 - B_1)(1 - B_2)(1 - B_3) \end{aligned} \qquad (13.1\text{-}2)$$

Example 13.5 Assume that $B_1 = .01$ and $B_2 = B_3 = .02$. Then the total blockage is

$$B_t = .01 + .0198 + .0194 = .0492 \approx 5\%$$

The result in Example 13.5 indicates that even when each individual blockage (i.e., B_1, B_2, and B_3) is small, the total blockage becomes very large. Therefore, the resultant blockage is what we are determining.

13.1.2 Call drops (dropped-call rate)

Call drops are defined as calls dropped for any reason after the voice channel has been assigned. Sometimes call drops due to weak signals are called *lost calls*. The dropped-call rate is partially based on the handoff-traffic model and partially based on signal coverage.

The handoff traffic model. A new handoff cell site treats handoffs the same way as it would an incoming call. Therefore, the blockage for handoff calls is also $B = .02$. Some MTSO systems may give priority to handoff calls rather than to incoming calls. In this case the blocking probability will be less than .02.

A warning feature can be implemented when the call cannot be handed off and may be dropped with high probability, enabling the customer to finish the call before it is dropped. Then the dropped-call rate can be reduced.

The loss of SAT calls. If the mobile unit does not receive a correct SAT in 5 s, the mobile-unit transmitter is shut down. If the mobile unit does not send back a SAT in 5 s, the transmitter at the cell site is shut down. In both cases the call is dropped. If the correct SAT cannot be detected at the cell site, as in cases of strong interference, then (1) the SAT can be offset by more than 15 Hz (see section entitled "The total dropped-call rate," below) or (2) the SAT tone generator in the mobile unit may not produce the desired tone.

Calculation of SAT interference conditions. The desired SAT is $\cos w_1 t$, and the undesired SAT is $\rho \cos w_2 t$. When $\rho \ll 1$, the SAT detector at the cell site can easily detect w_1. When ρ is greater and starts to approach 1, SAT interference occurs. The following analysis shows the

degree of the interference due to the value of ρ.

$$\cos w_1 t + \rho \cos w_2 t = A(t) \cos \theta(t) \tag{13.1-3}$$

where

$$A(t) = \sqrt{1 - \rho^2 + 2\rho \cos (w_1 - w_2)t} \tag{13.1-4}$$

$$\psi(t) = w_2 t - w_1 t$$

$$\theta = w_1 t + \tan^{-1} \frac{\rho \sin \psi(t)}{1 + \rho \cos \psi(t)} \tag{13.1-5}$$

$$w = \frac{d\theta}{dt} = w_1 + \frac{w_2 - w_1}{([1 + \rho \cos \psi(t)]/\{\rho[\rho + \cos \psi(t)]\}) + 1} \qquad (13.1\text{-}6)$$

Let $\cos \psi(t) = 1$ in Eq. (13.1-6), the extreme condition of w which is the offset frequency from a desired SAT.

$$w = w_1 + \left(1 + \frac{1}{\rho}\right)^{-1} (w_2 - w_1) \qquad (13.1\text{-}7)$$

For

$$\rho = \begin{cases} .3 & w = w_1 + .22(w_2 - w_1) \\ .5 & w = w_1 + .333(w_2 - w_1) \\ .75 & w = w_1 + .45(w_2 - w_1) \end{cases} \qquad (13.1\text{-}8)$$

If $w_2 - w_1 = 30$ Hz, for two adjacent SATs

$$\rho = \begin{cases} .5 & \omega = \omega_1 \pm 9.95 \quad \text{(acceptable)} \\ .75 & \omega = \omega_1 \pm 12.9 \quad \text{(marginal)} \end{cases}$$

If $\omega_2 - \omega_1 = 60$ Hz, for two ends of SATs

$$\rho = \begin{cases} .3 & \omega = \omega_1 \pm 13.8 \quad \text{(marginal)} \\ .5 & \omega = \omega_1 \pm 19.9 \quad \text{(unacceptable)} \\ .75 & \omega = \omega_1 \pm 25.8 \quad \text{(unacceptable)} \end{cases}$$

Adjacent SATs cannot interfere with the desired SAT for an undesired SAT level below $\rho = .75$ (meaning a level of -2.5 dB). However, when two SATs which are not adjacent to each other, the undesired SAT level should at least be lower than $\rho = 0.3$ (-10.5 dB) in order for no interference to occur.

Unsuccessful complete handoffs. Because of the limitations in processor capacity, the duration of the handoff process may occasionally be too long, and the mobile unit may not be informed of a new channel to be handed off.

The total dropped-call rate. Assume that the handoff blocking is B_4, the probability of lost SAT calls is B_5, and the probability of an unsuccessful complete handoff is B_6. Then the total drop call rate is

$$B_d = B_4 + B_5 (1 - B_4) + B_6 (1 - B_4)(1 - B_5) \qquad (13.1\text{-}9)$$

Usually, the dropped-call rate should be less than 5 percent.

13.2 Signaling Evaluation

The signaling protocols of existing systems are evaluated in this section. The signaling format of the forward control channel (FOCC) as deduced from a BCH code (63, 51) becomes a short code of (40, 28) or that of the reverse control channel (RECC) from a BCH becomes a short code of (48, 36), as described in Chap. 3. The 12 parity-check bits always remain unchanged. This BCH code can correct one error and detect two errors.

13.2.1 False-alarm rate

The false-alarm rate is the rate of occurrence of a false recognizable word that would cause a malfunction in a system. The false-alarm rate

should be less than 10^{-7}. Now we would like to verify that the BCH code (40, 28) can meet this requirement. The Hamming distance d of BCH (40, 28) is 5. This means that in every different code word at least 5 out of 40 bits are different. Then the false-alarm rate FAR can be calculated as

$$FAR = p_e^d(1 - p_e)^{L-d} \tag{13.2-1}$$

where p_e is the BER and d is the length of a word in bits.

Assume that in a noncoherent frequency-shift-keying (FSK) modulation system, the average BER of a data stream in a Rayleigh fading environment is

$$\langle p_e \rangle = \frac{1}{2 + \Gamma} \qquad \text{(noncoherent FSK)} \tag{13.2-2a}$$

and the average BER of a data stream received by a differential phase-shift-keying (DPSK) modulation system in the same environment is

$$\langle p_e \rangle = \tfrac{1}{2}\left(\frac{1}{\Gamma + 1}\right) \qquad \text{(DPSK)} \tag{13.2-2b}$$

where Γ is the carrier-to-noise ratio. Let* $\Gamma = 15$ dB; then we obtain BER from Eq. (13.2-2) as

$$\langle p_e \rangle = .03 \qquad \text{(noncoherent FSK)}$$

$$\langle p_e \rangle = .015 \qquad \text{(DPSK)}$$

Substituting $p_e = .03$, which is the higher BER, into Eq. (13.2-1), we obtain

$$FAR \approx (.03)^5 = 2.43 \times 10^{-8} \leq 10^{-7}$$

This meets the requirement that $FAR < 10^{-7}$.

13.2.2 Word error rate consideration

The word error rate (WER) plays an important role in a Rayleigh fading environment. The length of a word of an FOCC is $L = 40$ and the transmission rate is 10 kbps. Then the transmission time for a 40-bit word is

$$T = \frac{40}{10{,}000} = 4 \text{ ms}$$

* The C/N ratio of a data channel can be lower than 18 dB of a voice channel.

From Sec. 1.6.6, the average duration of fades can be obtained from the following assumptions: frequency = 850 MHz, vehicle speed = 15 mi/h, and threshold level = −10 dB (10 dB below the average power level); then the average duration of fades is

$$\bar{t} = 0.33 \times \left(\sqrt{2\pi}\,\frac{V}{\lambda}\right)^{-1} = 7 \text{ ms} \qquad (13.2\text{-}3)$$

Equation (13.2-3) shows that the transmission time of one word is shorter than the average duration of fades while the vehicle speed is 15 mi/h; that is, the whole word can disappear under the fade. Therefore, redundancy schemes are introduced. From Chap. 3, the FOCC format is

200 bits	word A (40 bits × 5 times), 28 information bits
200 bits	word B (40 bits × 5 times), 28 information bits
10 bits	bit synchronization
11 bits	word synchronization
42 bits	Busy/Idle-status bits
463 bits	56 information bits

The throughput can be obtained from

$$\frac{56}{463} = \frac{1200 \text{ bps}}{10{,}000 \text{ bps}}$$

Therefore, the throughput is 1200 bps (baseband rate).

13.2.3 Word error rate calculation

The WER can be calculated as follows. We may use a DPSK system because it has a general but simple analytic formula, more general than Eq. (13.2-2b)

$$\langle p_e \rangle = \tfrac{1}{2}\left(\frac{1}{\Gamma + 1}\right)^{M} \qquad (13.2\text{-}4)$$

where M is the number of diversity branches. It is difficult to obtain the WER from the correlation coefficient in the bit stream at a specific vehicle speed because the correlation coefficients of any two bits among all the bits in a word at that particular speed form a correlation coefficient matrix which is difficult to handle. Fortunately, we can find two extreme values, one at the speed $V \to \infty$ and the other at the speed $V - \to 0$. The calculations are described in detail in Ref. 3. Here we are simply illustrating the results.[4]

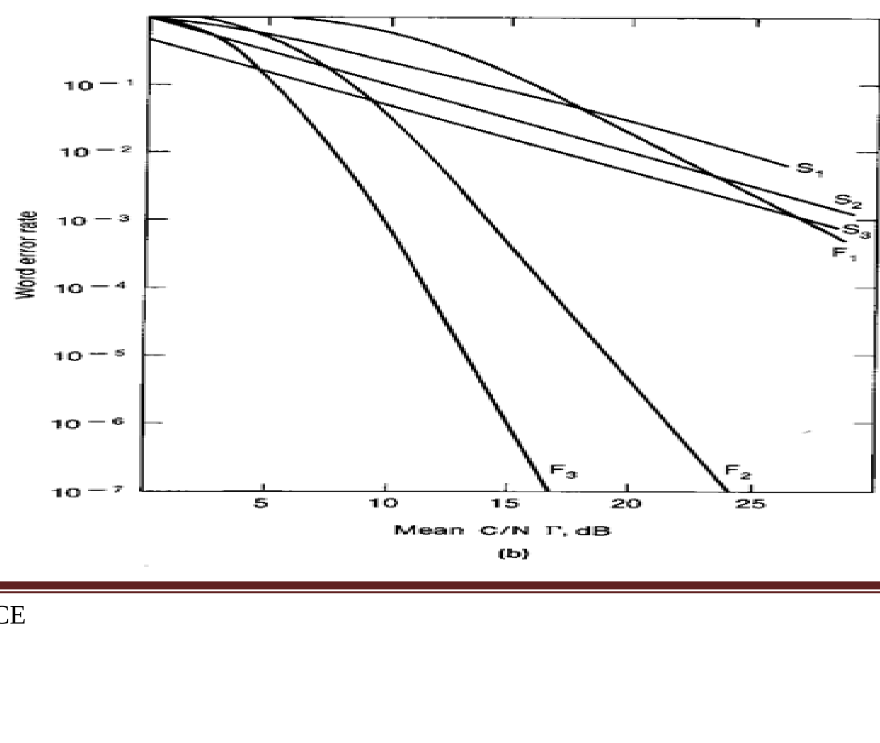
The performance of word error rates is shown in Fig. 13.2 for two cases: (1) no error correction and (2) one error correction. We have noticed that without redundancy (no repeat), the WER of a fast-fading case is worse than that of a slow-fading case. The WER obtained from a finite speed will lie between these two curves.

13.3 Measurement of Average Received Level and Level Crossings

13.3.1 Calculating average signal strength[5]

The signal strength can be averaged properly to represent a true local mean $m(x)$ to eliminate the Rayleigh fluctuation and retain the long-term fading information due to the terrain configuration. Let $\hat{m}(x)$ be the estimated local mean. If a length of data L is chosen properly, $\hat{m}(x)$ will approach $m(x)$ as

$$\hat{m}(x) = \frac{1}{2L}\int_{x-L}^{x+L} r(y)\,dy = \frac{1}{2L}\int_{x-L}^{x+L} m(y)r_0(y)\,dy$$
$$= m(x)\left[\frac{1}{2L}\int_{x-L}^{x+L} r_0(y)\,dy\right] = m(x) \qquad (13.3\text{-}1)$$



(b)

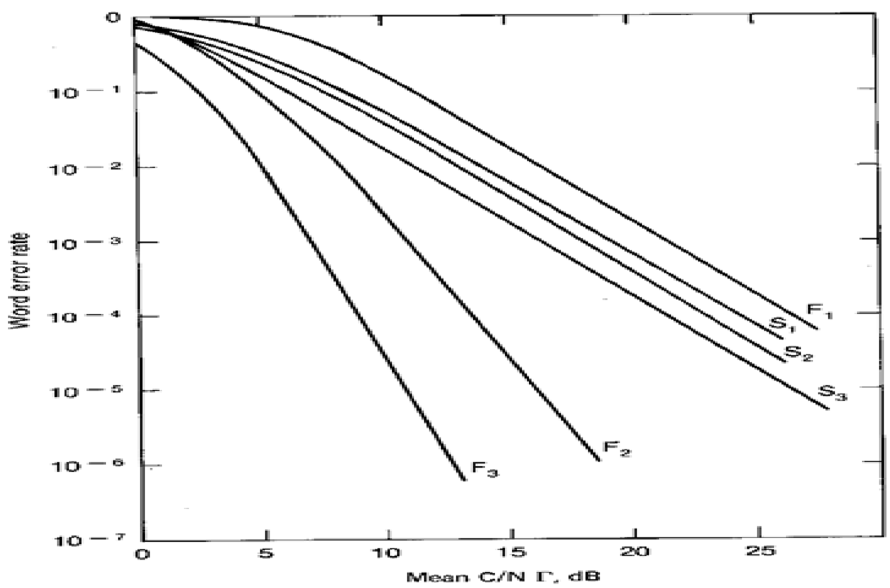
or

$$\frac{1}{2L}\int_{x-L}^{x+L} r_0(y)\,dy \rightarrow 1 \qquad (13.3\text{-}2)$$

where $r_0(y)$ is a Rayleigh distributed variable. If the value of Eq. (13.3-2) is close to 1, then $\hat{m}(x)$ is close to $m(x)$. The spread of $\hat{m}(x)$, denoted as $\sigma_{\hat{m}}$, can be expressed as

$$1\sigma_{\hat{m}} \text{ spread} = 20 \log \frac{m(x) + \sigma_{\hat{m}}}{m(x) - \sigma_{\hat{m}}} \quad \text{in dB} \qquad (13.3\text{-}3)$$

Equation (13.3-3) is plotted in Fig. 13.4. The $1\sigma_{\hat{m}}$ spread is used to indicate the uncertainty range of a measured mean value from a true mean value if the length of the data record is inadequate.



(a)

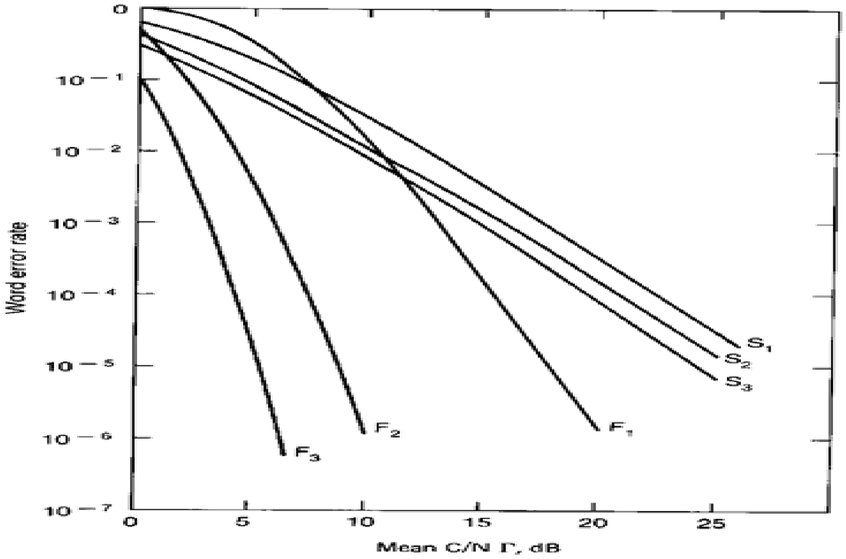
Figure 13.3 Word error rate for $N = 40$ bits. Number of branches $M = 2$; S_1, F_1 no repeat ($K = 1$); S_2, F_2 two-thirds voting

The proper length 2*L*. If we are willing to tolerate $1\sigma_{\hat{m}}$ spread in a range of 1.56 dB, then $2L = 20\lambda$. If the tolerated spread is in a range of 1 dB, then $2L = 40\lambda$.

For length $2L$ less than 20 wavelengths, the $1\sigma_{\hat{m}}$ spread begins to increase quickly. When length $2L$ is greater than 40λ, the $1\sigma_{\hat{m}}$ spread decreases very slowly.

In addition, the mobile radio signal contains two kinds of statistical distributions: $m(y)$ and $r_0(y)$. If a piece of signal data $r(y)$ is averaged, we find that if the length is shorter than 40λ, the unwanted $r_0(y)$ may be retained whereas at lengths above 40λ smoothing out of long-term fading $m(y)$ information may result. Therefore, 20 to 40λ is the proper length for averaging the Rayleigh fading signal $r(y)$.

Sampling average.* As mentioned previously, when using the averaging process with a filter, it is difficult to control bandwidth even when the length of the data to be integrated is appropriate. Therefore,



(b)

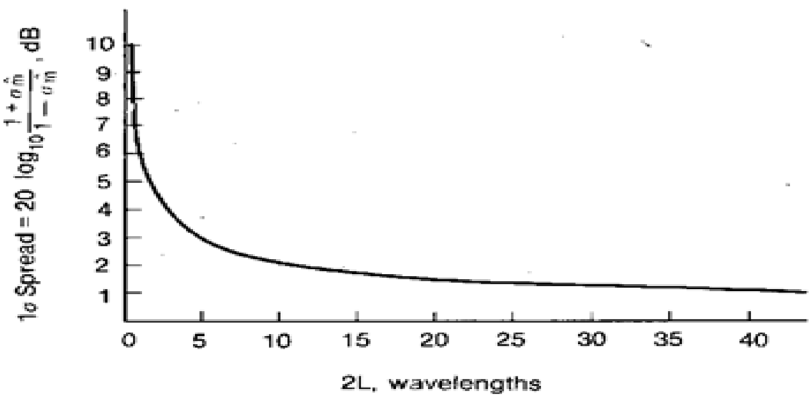


Figure 13.4 The value of $1\sigma_{\hat{m}}$ spread.

the sample values of $r(t)$ are used for sampling averaging instead of analog (continuous waveform) averaging. Then we must determine how many samples need to be digitized across a signal length of $2L$ (see Fig. 13.5). The number of samples taken for averaging should be as small as possible. However, we have to calculate how many sample points are needed for adequate results. We set a confidence level of 90 percent and determine the number of samples required for the sampling average. The general formula is

$$P\left(-1.65 \leq \frac{\bar{r}_j - \hat{m}_j}{\hat{\sigma}_j} \leq 1.65\right) = 90\% \qquad (13.3\text{-}4)$$

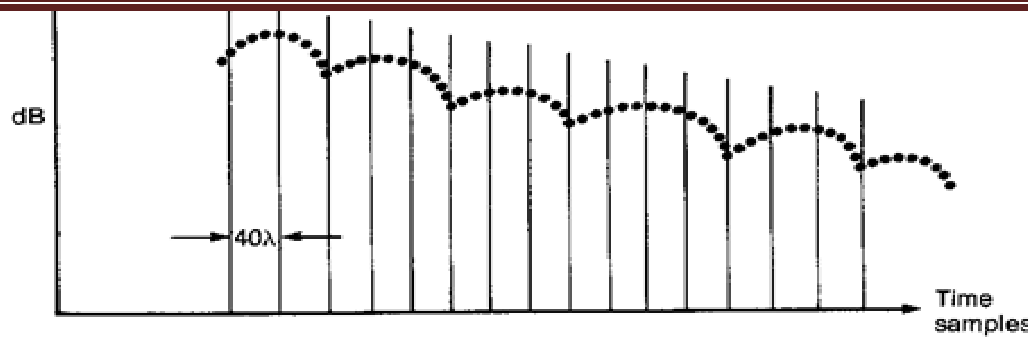


Figure 13.5 Sample average over a $2L = 40\ \lambda$ of data.

* The detailed derivation is shown in W. C. Y. Lee, *Mobile Communications Design Fundamentals,* Howard W. Sams & Co., 1986, Sec. 2.2.2.

Let $\hat{m}_j$ and $\hat{\sigma}_j$ be the mean and the standard deviation of ensemble average* $\bar{r}_j$ of jth interval ($2L$) and r_j be a gaussian variable

$$\hat{\sigma} = \frac{\sigma_r}{N} \qquad \hat{m} = m$$

where m and σ_r are the mean and the standard deviation of a Rayleigh sample r. N is the number of samples. Therefore,[5]

$$m = \frac{\sqrt{\pi}}{2}\sqrt{\overline{r^2}} \tag{13.3-5}$$

$$\sigma_r = \frac{\sqrt{4-\pi}}{2}\sqrt{\overline{r^2}} \tag{13.3-6}$$

$$\frac{\sigma_r}{m} = \sqrt{\frac{4-\pi}{\pi}} \tag{13.3-7}$$

Substitution of Eqs. (13.3-5) to (13.3-7) into Eq. (13.3-4) yields

$$P\left(\left(1 - \frac{0.8625}{\sqrt{N}}\right) m \le \bar{r}_j \le \left(1 + \frac{0.8625}{\sqrt{N}}\right) m\right) = 90\% \tag{13.3-8}$$

Then the 90 percent confidence interval CI expressed in decibels is

$$90\%\ CI = 20 \log\left(1 + \frac{0.8625}{\sqrt{N}}\right)$$

$$N = \begin{cases} 50 & 90\%\ CI = 1\text{ dB} \\ 36 & 90\%\ CI = 1.17\text{ dB} \end{cases} \tag{13.3-9}$$

13.3.2 Estimating unbiased average noise levels[6]

Usually the sampled noise in a mobile environment contains high-level impulses that are generated by the ignition noise of the gasoline

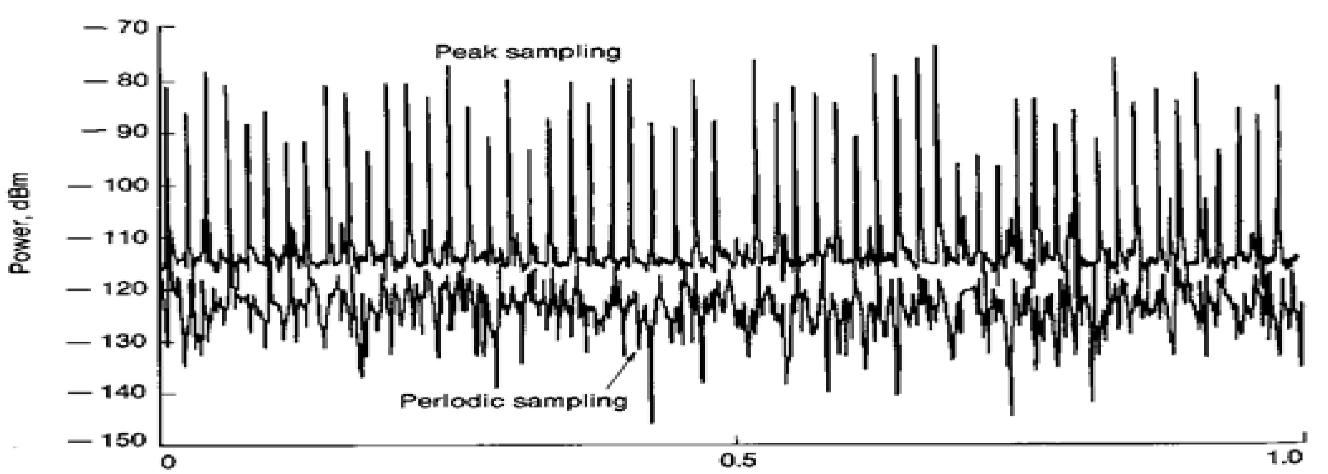
* Time average in a mobile radio environment is an ergodic process in statistics. Therefore the values from a time average with a proper interval and an ensemble average are the same.

engine. Although the level of these impulses is high, the pulse width of each impulse generally is very narrow (see Fig. 13.6). As a result, the energy contained in each impulse is very small and should not have any noticeable effect on changing the average power in a 0.5 s interval.

However, in a normal situation averaging a sampled noise is done by adding up the power values of all samples, including the impulse samples, and dividing the sum by the number of samples. This is called the *conventionally averaged noise power* and is denoted n_c. In this case, BER and signal-to-noise ratio measured in certain geographic areas. In a new statistical method the average noise is estimated by excluding the noise impulses while retaining other forms of interference. This technique is compatible with real-time processing constraints.

Description of the method. A counter in the mobile unit counts the instantaneous noise measurements which fall below a preset threshold level X_t and sends a message containing the number of counts n to the database for recording. From the database data, we can calculate the percentage of noise samples x_i below the present level X_t

$$P(x_i \leq X_t) = \frac{n}{N} \tag{13.3-10}$$



13.3.3 Signal-strength conversion

Confusion arises because the field strength (in decibels above 1 μV, dBμ) is measured in free space, and the power level in decibels above 1 mW (dBm) is measured at the terminal impedance of a given receiving antenna. Furthermore, the dimensions of the two units are different. The signal field strength measured on a linear scale is in microvolts per meter (μV/m), and the power level measured on a linear scale is in milliwatts (or watts).

Further confusion arises because of the notation "dBμ." Sometimes dBμ means the number of decibels above 1 μV measured at a given voltage. Sometimes, it represents the number of decibels referred to microvolts per meter when field strength is being measured.

The conversion from decibels (microvolts per meter, dBμ) to decibels above 1 mW (dBm) at 850 MHz is shown in Eq. (5.1-13) using the relationship between induced voltage and effective antenna length.[7] The conversion at a frequency other than 850 MHz can be obtained as follows.[8]

$$P_{max}(\text{in dBm at } f_1 \text{ MHz}) = P_{max}(\text{in dBm at 850 MHz}) + 20 \log \left(\frac{850}{f_1}\right) \qquad (13.3\text{-}12)$$

where f_1 is in megahertz. The details of this conversion are given in Sec. 5.1.3.

13.3.4 Receiver sensitivity

The sensitivity of a radio receiver is a measure of its ability to receive weak signals. The sensitivity can be expressed in microvolts or in decibels above 1 μV.

$$Y \quad \text{dB}\mu\text{V} = 20 \log (x \quad \mu\text{V}) \qquad (13.3\text{-}13)$$

Also, the sensitivity can be expressed in milliwatts or dBm.

$$y \quad \text{dBm} = 10 \log (x \quad \text{mW}) \qquad (13.3\text{-}14)$$

The conversion from microvolts to decibels above 1 mW, assuming a 50-Ω terminal, has been shown in Eq. (5.1-15) as

$$0 \text{ dB}\mu\text{V} = 10 \log \frac{(1 \times 10^{-6})^2}{50} = -137 \text{ dBW} = -107 \text{ dBm} \qquad (13.3\text{-}15)$$

13.3.5 Level-crossing counter[9]

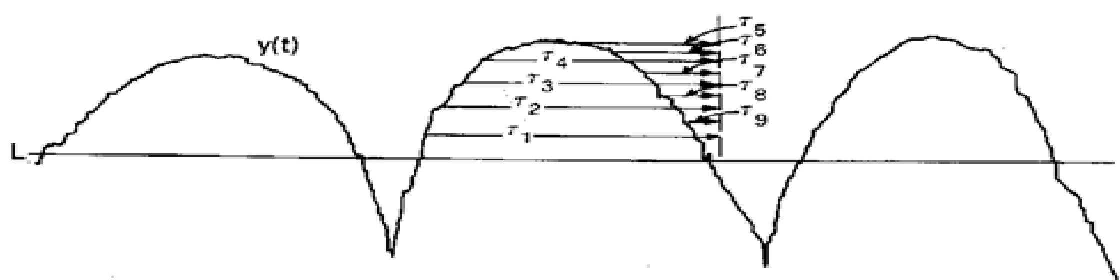
A signal fading level crossing counter will face a false-count problem as a result of the granular noise as shown in Fig. 13.9. The positive slope crossing count should be 3, but the false counts may be 12. A proposed level-crossing counter can eliminate the false counts. First, by sampling the fading signal at an interval of T seconds, we can choose the interval T such that $1/T$ is small in comparison to the fading rate. The duration of stay τ_i is measured for every sample time which is above level L and the time span until the signal drops below level L. We can also use a device for measuring the percentage of time that $y(t)$ is above L. This device may be called a level crossing counter.

Let us define

$$p = P_r[y(t) \geq L] \tag{13.3-16}$$

and

$$q = 1 - p = P[y(t) < L] \tag{13.3-17}$$



We count the number of times M that τ_i is above L and then sum the duration of total stays $T_s (T_s = \Sigma_{i=1}^{M} \tau_i)$ above L. The average duration of upward fading is

$$\tau_p = \frac{2T_s}{M} \tag{13.3-18}$$

The average duration of fades τ_q where $y(t)$ is below L is

$$\tau_q = \tau_p \frac{1 - p}{p} \tag{13.3-19}$$

and the level crossing rate n at level L is

$$n = \frac{1}{\tau_p + \tau_q} = \frac{p}{\tau_p} \tag{13.3-20}$$

The advantage of this method is that we stop our time count whenever $y(t)$ crosses L, thus avoiding false counts due to noise when $y(t)$ is close to level L. Obviously noise can give an incorrect measure of the "duration of stay" for a single interval; however, noise shortens many intervals while it lengthens others, and thus, when averaged over many cycles, the "duration of stay" is an accurate number.