MODULE IV

MULTIPLEXING

Multiplexing describes how several sources can share a medium with minimum or no interference.
Example: Highways with several lanes. Many users (car drivers) use the same medium (the highways) with hopefully no interference (i.e., accidents).

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There are n inputs to a multiplexer. The multiplexer is connected by a single data link to a demultiplexer. The link is able to carry n separate channels of data. The multiplexer combines (multiplexes) data from the n input lines and transmits over a higher capacity data link. The demultiplexer accepts the multiplexed data stream, separates (demultiplexes) the data according to channel, and delivers data to the appropriate output lines.

For wireless communication, multiplexing can be carried out in four dimensions: **space**, **frequency**, **time** and **code**. In this field, the task of multiplexing is to assign space, frequency, time and code to each communication channel with a minimum of interference and a maximum of medium utilization.

MULTIPLEXING TECHNIQUES:

1. Space division multiplexing (SDM)
2. Frequency division multiplexing (FDM)
3. Time division multiplexing (TDM) or Synchronous TDM
4. Asynchronous TDM or Statistical TDM or Intelligent TDM
5. Code division Multiplexing (CDM)

1. SPACE DIVISION MULTIPLEXING (SDM)

- The task of multiplexing is to assign space to each communication channel with a minimum of interference and a maximum of medium utilization.
- Used in FM radio stations.

Figure shows six channels k_i and introduces a three dimensional coordinate system. This system shows the dimensions of code c, time t and frequency f. For S**pace Division Multiplexing (SDM)**, the (three dimensional) space s_i is also shown. Here space is represented via circles indicating the interference range The channels k1 to k3 can be mapped onto the three spaces s1 to s3 which clearly separate the channels and prevent the interference ranges from overlapping. The space between the interference ranges is called **guard space**. For the remaining channels (k4 to k6) three additional spaces would be needed.

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In wireless transmission**, SDM implies a separate sender for each communication channel with a wide enough distance between senders**. This multiplexing scheme is used, for example, at FM radio stations where the transmission range is limited to a certain region – many radio stations around the world can use the same frequency without interference. Using SDM, **problems** arise if two or more channels were established within the same space, for example, if several radio stations want to broadcast in the same city. Then, one of the following multiplexing schemes must be used (frequency, time, or code division multiplexing).

2. FREQUENCY DIVISION MULTIPLEXING (FDM)

Characteristics:

- A number of signals can be carried simultaneously if each signal is modulated onto a different carrier frequency
- The carrier frequencies are sufficiently separated that the bandwidths of the signals do not significantly overlap.
- FDM is possible when the useful bandwidth of the transmission medium exceeds the required bandwidth of signals to be transmitted.
- Used in radio or television set.

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- Six signal sources are fed into a multiplexer, which modulates each signal onto a different frequency (f1,…. , f6).
- Each modulated signal requires a certain bandwidth centered on its carrier frequency, known as **channel**.
- To prevent interference, the channels are separated by **guard bands**, which are unused portions of the spectrum.
- The composite signal transmitted across the medium is analog.
- In the case of digital input, the input signals must be passed through modems to be converted to analog.

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Fig 4: A generic depiction of an FDM system. A number of analog or digital signals [$m_i(t)$, i=1,n] are to be multiplexed onto the same transmission medium. Each signal $m_i(t)$ is modulated onto a carrier f_i because multiple carriers are to be used, each is referred to as a **subcarrier**. Any type of modulation may be used. The resulting analog, modulated signals are then summed to produce a composite baseband signal $m_b(t)$ (Figure 8.4b). The spectrum of signal $m_i(t)$ is shifted to be centered on f_i. For this scheme to work, f_i must be chosen so that the bandwidths of the various signals do not significantly overlap. Otherwise, it will be impossible to recover the original signals. The composite signal may then be shifted as a whole to another carrier frequency by an additional modulation step.

The FDM signal s(t) has a **total bandwidth** B, where $B > \Sigma B_i$ .This analog signal may be transmitted over a suitable medium. At the receiving end, the FDM signal is demodulated to retrieve $m_b(t)$ which is then passed through n bandpass filters, each filter centered on f_i and having a bandwidth B_i for $1 <= i <= n$. In this way, the signal is again split into its component parts. Each component is then demodulated to recover the original signal.

EXAMPLE: Consider a simple example of transmitting three voice signals simultaneously over a medium. The bandwidth of a voice signal is generally taken to be 4 kHz, with an effective spectrum of 300 to 3400 Hz (Figure 5a). If such a signal is used to amplitude-modulate a 64-kHz carrier, the spectrum of Figure 5b results. The modulated signal has a bandwidth of 8 kHz, extending from 60 to 68 kHz. To make efficient use of bandwidth, we elect to transmit only the lower sideband. If three voice signals are used to modulate carriers at 64, 68, and 72 kHz, and only the lower sideband of each is taken, the spectrum of Figure 5c results.

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Figure 5 points out two **problems** that an FDM system must cope with.

1. The first is crosstalk, which may occur if the spectra of adjacent component signals overlap significantly. In the case of voice signals, with an effective bandwidth of only 3100 Hz (300 to 3400), a 4-kHz bandwidth is adequate. The spectra of signals produced by modems for voice band transmission also fit well in this bandwidth.
2. Another potential problem is intermodulation noise. On a long link, the nonlinear effects of amplifiers on a signal in one channel could produce frequency components in other channels.

WAVELENGTH DIVISION MULTIPLEXING (WDM):

- Optical fiber is fully exploited when multiple beams of light at different frequencies are transmitted on the same fiber. This is a form of frequency division multiplexing (FDM) but is commonly called **wavelength division multiplexing** (WDM).
- With WDM, the light streaming through the fiber consists of many colors, or wavelengths, each carrying a separate channel of data.

- A typical WDM system has the same general architecture as other FDM systems.
- A number of sources generate a laser beam at different wavelengths.
- These are sent to a multiplexer, which consolidates the sources for transmission over a single fiber line.
- Optical amplifiers, spaced tens of kilometers apart, amplify all of the wavelengths simultaneously.
- Finally, the composite signal arrives at a demultiplexer, where the component channels are separated and sent to receivers at the destination point.

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Fig: Fiber-optical line using wavelength division multiplexing and supporting multiple-speed transmissions

- Most WDM systems operate in the 1550-nm range.
- In early systems, 200 GHz was allocated to each channel, but today most WDM systems use 50-GHz spacing.
- **Dense wavelength division multiplexing (DWDM)** use more channels, more closely spaced, than ordinary WDM. A channel spacing of 200 GHz or less could be considered dense.
- **Coarse wavelength division multiplexing (CWDM)** used for short distance connections, not closely spaced.

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3. TIME DIVISION MULTIPLEXING (TDM) OR SYNCHRONOUS TDM

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(a)Characteristics:

- Multiple digital signals (or analog signals carrying digital data) can be carried on a single transmission path by interleaving portions of each signal in time.
- The interleaving can be at the bit level or in blocks of bytes or larger quantities.
- Synchronous time division multiplexing is possible when the achievable data rate (bandwidth) of the medium exceeds the data rate of digital signals to be transmitted.
- Used for multiplexing digitized voice streams and data streams.

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Figure 6: A number of signals [$m_i(t) = i,n$] are to be multiplexed onto the same transmission medium. The signals carry digital data and are generally digital signals. The incoming data from each source are briefly buffered. Each buffer is typically one bit or one character in length. The buffers are scanned sequentially to form a composite digital data stream $m_c(t)$. The scan operation is sufficiently rapid so that each buffer is emptied before more data can arrive. Thus, the data rate of $m_c(t)$ must at least equal the sum of the data rates of the $m_i(t)$. The digital signal $m_c(t)$ may be transmitted directly, or passed through a modem so that an analog signal is transmitted. In either case, transmission is typically synchronous.

Figure 6b: The transmitted data format. The data are organized into **frames**. Each frame contains a cycle of time slots. In each frame, one or more slots are dedicated to each data source. The sequence of slots dedicated to one source, from frame to frame, is called a **channel**. The slot length equals the transmitter buffer length, typically a bit or a byte (character).

The **byte-interleaving technique** is used with asynchronous and synchronous sources. Each time slot contains one character of data. Typically, the start and stop bits of each character are eliminated before transmission and reinserted by the receiver, thus improving efficiency.

The **bit-interleaving technique** is used with synchronous sources and may also be used with asynchronous sources. Each time slot contains just one bit.

Fig 6c: At the **receiver**, the interleaved data are demultiplexed and routed to the appropriate destination buffer. For each input source $m_i(t)$ there is an identical output destination that will receive the output data at the same rate at which it was generated.

Synchronous TDM is called synchronous not because `**synchronous** transmission is used, but because the **time slots are preassigned** to sources and fixed.

(b)Digital Carrier Systems:

The long-distance carrier system was designed to transmit voice signals over high-capacity transmission links, such as optical fiber, coaxial cable, and microwave**.**

Figure 9:

- The basis of the TDM hierarchy is the **DS-1 transmission format** which multiplexes 24 channels.
- Each frame contains 8 bits per channel plus a framing bit for 24 x 8 + 1 = 193 bits.

For **voice transmission**,

- Each channel contains one word of digitized voice data.
- The original analog voice signal is digitized using pulse code modulation (PCM) at a rate of 8000 samples per second.
- Therefore, each channel slot and hence each frame must repeat **8000** times per second.
- With a frame length of 193 bits, a **data rate** of 8000 x 193 = **1.544 Mbps.**
- For five of every six frames, **8-bit PCM** samples are used.
- For every sixth frame, each channel contains a **7-bit PCM** word plus a **signaling bit**.
- The signaling bits form a stream for each voice channel that contains network control and routing information.

For **digital data** service,

- For compatibility with voice, the same 1.544-Mbps data rate is used.
- **23 channels** of data are provided.
- The **twenty-fourth channel** position is reserved for a special sync byte, which allows faster and more reliable reframing following a framing error.
- Within each channel, **7 bits per frame** are used for data, with the **eighth bit** used to indicate whether the channel, for that frame, contains user data or system control data.
- With 7 bits per channel, and because each frame is repeated **8000** times per second, a **data rate** of **56 kbps** can be provided per channel.
- Lower data rates are provided using a technique known as **subrate multiplexing**.
- For this technique, an additional bit is robbed from each channel to indicate which subrate multiplexing rate is being provided.
- This leaves a **total capacity per channel** of 6 x 8000 = **48 kbps.**
- This capacity is used to multiplex five 9.6-kbps channels, ten 4.8-kbps channels, or twenty 2.4-kbps channels.

Finally, the **DS-1 format** can be used to carry a **mixture of voice and data channels**. In this case, all 24 channels are utilized; no sync byte is provided.

(c)SONET/SDH:

- **SONET (Synchronous Optical Network)** is an optical transmission interface proposed by BellCore and standardized by ANSI.
- A compatible version, referred to as **Synchronous Digital Hierarchy (SDH)**, has been published by ITUT in Recommendation G.707.
- SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber.

Signal Hierarchy

- The SONET specification defines a hierarchy of standardized digital data rates (Table 4).
- Multiple STS-1 signals can be combined to form an STS-N signal.
- For the ITU-T Synchronous Digital Hierarchy, the lowest rate is 155.52 Mbps, which is designated STM-1.This corresponds to SONET STS-3.

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Frame Format

- The basic SONET building block is the **STS-1 frame**, which consists of 810 octets and is transmitted once every 125μs, for an overall data rate of 51.84 Mbps **(Figure 10a).**
- The frame can logically be viewed as a **matrix of 9 rows of 90 octets each**, with transmission being one row at a time, from left to right and top to bottom.
- The first three columns (3 octets x 9 rows = 27 octets) of the frame are devoted to **overhead octets.**
- Nine octets are devoted to **section-related overhead** and 18 octets are devoted to **line overhead**.
- **Figure 10b** shows the general format for higher-rate frames, using the ITU-T designation.

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- **Figure 11a** shows the arrangement of **overhead octets**
- The remainder of the frame is **payload**. The payload includes a column of **path overhead**, which is not necessarily in the first available column position; the line overhead contains a pointer that indicates where the path overhead starts.
- **Figure 11b** shows the arrangement of **path overhead octets**
- **Table 5** defines the various **fields**.

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4.STATISTICAL TIME DIVISION MULTIPLEXING OR ASYNCHRONOUS TDM

(a)Characteristics:

- In a **synchronous time division multiplexer** many of the time slots in a frame are wasted.
- **Application** of a **synchronous TDM** involves linking a number of terminals to a shared computer port. Even if all terminals are actively in use, most of the time there is no data transfer at any particular terminal.
- An alternative to synchronous TDM is **statistical or asynchronous TDM**. The statistical multiplexer **dynamically allocates time slots on demand.**
- As with a synchronous TDM, the statistical multiplexer has a number of **I/O lines** on one side and a higher-speed **multiplexed line** on the other.
- Each I/O line has a **buffer** associated with it.
- There are n I/O lines, but only **k**, where k<n **time slots** available on the TDM frame.
- For **input**, the function of the multiplexer is to scan the input buffers, collecting data until a frame is filled, and then send the frame.
- On **output**, the multiplexer receives a frame and distributes the slots of data to the appropriate output buffers.
- Because statistical TDM takes advantage of the fact that the attached devices are not all transmitting all of the time, the **data rate** on the multiplexed line is **less** than the sum of the data rates of the attached devices.
- Thus, a statistical multiplexer can use a **lower data rate** to support as many devices as a synchronous multiplexer.
- Improve the efficiency of synchronous TDM by adding complexity to the multiplexer.

- **Figure 12** compares statistical and synchronous TDM.
- The figure depicts four data sources and shows the data produced in four time epochs (t_0, t_1, t_2, t_3).
- In the case of **synchronous multiplexer**, the multiplexer has an effective output rate of four times the data rate of any of the input devices. During each epoch, data are collected from all four sources and sent out. For example, in the first epoch, sources C and D produce no data. Thus, two of the four time slots transmitted by the multiplexer are empty.
- In contrast, the **statistical multiplexer** does not send empty slots if there are data to send. Thus, during the first epoch, only slots for A and B are sent. However, the positional significance of the slots is lost in this scheme. It is not known ahead of time which source's data will be in any particular slot. Because data arrive from and are distributed to I/O lines unpredictably, address information is required to assure proper delivery. Thus, there is **more overhead** per slot for statistical TDM because each slot carries an address as well as data.

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Fig 13: Frame structure

A statistical TDM system will use a synchronous protocol such as HDLC. Figure 13 shows two possible formats.

1. **Fig 13 a:** Only one source of **data** is included per frame. That source is identified by an **address**. The length of the data field is variable, and its end is marked by the end of the overall frame.
2. **Fig 13b:** A way to improve efficiency is to allow **multiple data** sources to be packaged in a **single frame**. Thus, the statistical TDM **subframe** consists of a sequence of **data fields**, each labeled with an **address** and a **length**.The address field can be reduced by using relative addressing

ws 274fig13

(b)Cable Modem:

- To support data transfer to and from a cable modem, a cable TV provider dedicates **two channels**, one for transmission in each direction.
- Each **channel** is shared by a number of **subscribers**.
- **Figure 16:** A form of statistical TDM is used.
- In the downstream direction, cable **headend** to subscriber, a cable scheduler delivers data in the form of small packets.
- Because the channel is shared by a number of subscribers, if more than one subscriber is active, each subscriber gets only a **fraction** of the downstream capacity.
- An individual cable modem subscriber may experience **access speeds** from 500 kbps to 1.5 Mbps or more.
- When a subscriber has data to transmit, it must first **request** time slots on the shared upstream channel.
- Each subscriber is given dedicated **time slots** for this request purpose.
- The headend scheduler **responds** to a request packet by sending back an assignment of future time slots to be used by this subscriber.
- Thus, **a number of subscribers can share the same upstream channel** without conflict.

ws 280 fig 16

To support both cable television programming and data channels, the cable **spectrum** is divided into **three ranges**, each of which is further divided into 6-MHz channels.

1. User to Network data (upstream): 5 – 40 MHz
2. Television delivery (downstream): 50 – 550 MHz
3. Network to User data (downstream): 550 – 750 MHz

Figure 17: Cable modem configuration at a residential or office location

- At the interface of the external cable, a one-to-two splitter enables the subscriber to continue to receive cable television service through numerous FDM 6-MHz channels, while simultaneously supporting data channels to one or more computers in a local area network.
- The inbound channel first goes through a radio frequency (RF) tuner that selects and demodulates the data channel down to a spectrum of 0 to 6 MHz.
- This channel provides a data stream encoded using 64-QAM (Quadratic Amplitude Modulation) or 256 – QAM.

- The QAM demodulator extracts the encoded data stream and converts it to a digital signal that it passes to the Media Access Control (MAC) module.
- In the outbound direction, a data stream is modulated using either QPSK (Quadrature Phase Shift Keying) or 16-QAM.

ws fig 17 281

5.CODE DIVISION MULTIPLEXING (CDM)

- CDM allows multiple users to share a common set of frequencies by assigning a unique digital code to each user.
- It is based upon a class of modulation techniques known as spread spectrum technology.
- Spread spectrum is a technique used in the communication industry for modulating a signal into a new signal that is more secure and thus more resistant to wiretapping.
- CDM uses Direct Sequence Spread Spectrum (DSSS) technology that spreads the transmission of a signal over a wide range of frequencies using mathematical values.
- As the original data is input into a direct sequence modulator, each binary 1 and 0 is replaced with a larger unique bit sequence.
- Used in military applications and cellular telephone companies.

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Figure above shows how all channels ki use the same frequency at the same time for transmission. Separation is now achieved by assigning each channel its own ‘code’, **guard spaces** are realized by using codes with the necessary ‘distance’ in code space, e.g., **orthogonal codes**.

Advantages:

- Good protection against interference and tapping.
- Different codes have to be assigned, but code space is huge compared to the frequency space.
- Assigning individual codes to each sender does not usually cause problems.

Disadvantages:

- Relatively high complexity of the receiver.
- A receiver has to know the code and must separate the channel with user data from the background noise composed of other signals and environmental noise.
- A receiver must be precisely synchronized with the transmitter to apply the decoding correctly.
- All signals should reach a receiver with almost equal strength, otherwise some signals could drain others.

MULTIPLE ACCESS - CODE DIVISION MULTIPLE ACCESS (CDMA)

- Codes with certain characteristics can be applied to the transmission to enable the use of **code division multiplexing (CDM)**.
- **Code division multiple access (CDMA)** systems use exactly these codes to separate different users in code space and to enable access to a shared medium without interference.
- The main **problem** is how to find "good" codes and how to separate the signal from noise generated by other signals and the environment.
- A code for a certain user should have a good **autocorrelation** and should be **orthogonal** to other codes.

Orthogonal

- Think of a system of coordinates and vectors starting at the origin, i.e., in (0, 0, 0).
- **Two vectors are called orthogonal if their inner product is 0**
- The two vectors (2, 5, 0) and (0, 0, 17) -> (2, 5, 0) * (0, 0, 17) = 0 + 0 + 0 = 0. -> orthogonal
- The two vectors (1, 2, 3) and (4, 2, –6) -> (1, 2, 3) * (4, 2, 6) = 4 +4 - 18 = -10 -> not orthogonal
- The two vectors (1, 2, 3) and (4, 2, –3) -> (1, 2, 3) * (4, 2, -3) = 4 + 4 - 9 = -1 -> almost orthogonal, (which is "close" to zero).

Autocorrelation

- **To have good autocorrelation, the inner product with itself should be large.**
- Example: The Barker code (+1, –1, +1, +1, –1, +1, +1, +1, –1, –1, –1)
  Solution: (+1, –1, +1, +1, –1, +1, +1, +1, –1, –1, –1) * (+1, –1, +1, +1, –1, +1, +1, +1, –1, –1, –1)
  = (1+1+1+1+1+1+1+1+1+1+1)
  =11
  11 is large value, therefore good correlation.

Example explains the basic function of CDMA before it is applied to signals:

- Two senders, A and B, want to send data. CDMA assigns the following unique and orthogonal key sequences: key Ak = 010011 for sender A, key BK = 110101 for sender B. Sender A wants to send the bit Ad = 1, sender B sends Bd = 0. To illustrate this example, let us assume that we code a binary 0 as –1, a binary 1 as +1. We can then apply the standard addition and multiplication rules.

- Both senders spread their signal using their key as chipping sequence (the term ‘spreading’ here refers to the simple multiplication of the data bit with the whole chipping sequence). In reality, parts of a much longer chipping sequence are applied to single bits for spreading. Sender A then sends the signal As = Ad*Ak = +1*(–1, +1, –1, –1, +1, +1) = (–1, +1, –1, –1, +1, +1). Sender B does the same with its data to spread the signal with the code: Bs = Bd*Bk = –1*(+1, +1, –1, +1, –1, +1) = (–1, –1, +1, –1, +1, –1).
- Both signals are then transmitted at the same time using the same frequency, so, the signals superimpose in space (analog modulation is neglected in this example). Discounting interference from other senders and environmental noise from this simple example, and assuming that the signals have the same strength at the receiver, the following signal C is received at a receiver: C = As + Bs = (–2, 0, 0, –2, +2, 0).
- The receiver now wants to receive data from sender A and, therefore, tunes in to the code of A, i.e., applies A’s code for despreading: C*Ak = (–2, 0, 0, –2, +2, 0)*(–1, +1, –1, –1, +1, +1) = 2 + 0 + 0 + 2 + 2 + 0 = 6. As the result is much larger than 0, the receiver detects a binary 1. Tuning in to sender B, i.e., applying B’s code gives C*Bk = (–2, 0, 0, –2, +2, 0)* (+1, +1, –1, +1, –1, +1) = –2 + 0 + 0 – 2 – 2 + 0 = –6. The result is negative, so a 0 has been detected.

sch 85 14 15

Figure 3.14 shows a sender A that wants to transmit the bits 101. The key of A is shown as signal and binary key sequence Ak. In this example, the binary “0” is assigned a positive signal value, the binary “1” a negative signal value. After spreading, i.e., XORing Ad and Ak, the resulting signal is As.

The same happens with data from sender B, here the bits are 100. The result of spreading with the code is the signal Bs. As and Bs now superimpose during transmission (again without noise and both signals having the same strength). The resulting signal is simply the sum As + Bs as shown in **Figure 3.15.**

A receiver now tries to reconstruct the original data from A, Ad. Therefore the receiver applies A's key, Ak, to the received signal and feeds the result into an integrator (see section 2.7.1). The integrator adds the products (i.e., calculates the inner product), a comparator then has to decide if the result is a 0 or a 1 as shown in **Figure 3.16.** As we can see, although the original signal form is distorted by B's signal, the result is still quite clear.

The same happens if a receiver wants to receive B's data (see **Figure 3.17**). The comparator can easily detect the original data. Looking at (As + Bs)*Bk one can also imagine what could happen if A's signal was much stronger and noise distorted the signal. The little peaks which are now caused by A's signal would be much higher, and the result of the integrator would be wrong. If Ak and Bk are perfectly orthogonal and no noise disturbs the transmission, the method works (in theory) for arbitrarily different signal strengths.

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Finally**, Figure 3.18** shows what happens if a receiver has the wrong key or is not synchronized with the chipping sequence of the transmitter. The integrator still presents a value after each bit period, but now it is not always possible for the comparator to decide for a 1 or a 0, as the signal rather resembles noise. Integrating over noise results in values close to zero. Even if the comparator could detect a clear 1, this could still not reconstruct the whole bit sequence transmitted by a sender. A checksum on layer 2 would detect the erroneous packet. This illustrates CDMA's inherent protection against tapping. It is also the reason for calling the spreading code a key, as it is simultaneously used for encryption on the physical layer.

sch 87 3.18

SPREAD ALOHA MULTIPLE ACCESS (SAMA):

- If we use CDMA with only a single code, the resulting scheme is called **Spread Aloha Multiple Access (SAMA)** and is a combination of CDMA and TDMA

SAMA works as follows: Each sender uses the same spreading code (**Figure 3.19** this is the code 110101). The standard case for Aloha access is shown in the upper part of the figure. Sender A and sender B access the medium at the same time in their narrowband spectrum, so that all three bits shown cause a collision.

The same data could also be sent with higher power for a shorter period as shown in the middle, but now spread spectrum is used to spread the shorter signals, i.e., to increase the bandwidth (spreading factor s = 6 in the example). Both signals are spread, but the chipping phase differs slightly. Separation of the two signals is still possible if one receiver is synchronized to sender A and another one to sender B. The signal of an unsynchronized sender appears as noise. The probability of a ‘collision’ is quite low if the number of simultaneous transmitters stays below 0.1 0.2s. This also depends on the noise level of the environment.

Advantages:

- Robustness against narrowband interference
- Simple coexistence with other systems in the same frequency bands.

Disadvantage:

- Finding good chipping sequences
- Code is not orthogonal to itself – it should have a good autocorrelation but, at the same time, correlation should be low if the phase differs slightly.