Advanced Topics in Communications II (ELCN 456)

(C) Samy S. Soliman

EECE Department - Cairo University, Egypt Zewail City - University of Science and Technology, Egypt

Email: samy.soliman@cu.edu.eg Website: http://scholar.cu.edu.eg/samysoliman

Credit Hours System

Transmission Systems Transmission Impairments

- The basic building block for transmission is the telephone channel or voice channel.
- Voice channel implies spectral occupancy
- There are three basic impairments regarding the voice channel
	- Attenuation distortion
	- **Phase distortion**
	- Noise
- Additionally, there are
	- Echo
	- Singing
	- Level

- Signal over a voice channel suffers from distortion Output signal is not an exact replica of the input
- One form of distortion is attenuation distortion
	- Imperfect amplitude-frequency response
	- Can be avoided if all frequencies within the bandpass have exactly same loss (or gain)
	- In reality, some frequencies are attenuated more than others
	- For example, on loaded wire-pair systems, higher frequencies are attenuated more

1- Attenuation Distortion

←□

- Non-linear phase shift
- **Delay:** finite time a signal takes to pass the total extension of a voice channel or any network
- **•** Propagation time is different for different frequencies
- Wavefront of one frequency arriving before the wavefront of another in the passband
- Modulated signal will not be distorted if the phase shift changes uniformly with frequency
- If the phase shift is nonlinear with respect to frequency, the output signal is distorted compared to the input

2- Phase Distortion

- Velocity of propagation varies with frequency because of the electrical characteristics associated with the network
- In a voice channel, velocity of propagation increases toward band center and decreases toward band edge

つひひ

2- Phase Distortion: EED

- If the phase-frequency relationship over a band is not linear, distortion will occur in the transmitted signal
- This phase distortion is often measured by **envelope delay distortion** (EDD), which occurs when the rate of change of phase shift with frequency over the necessary BW is not constant
- Mathematically, envelope delay is the derivative of the phase shift with respect to frequency
- EDD equals the maximum difference in the derivative over any frequency interval
- EDD is always a difference between the envelope delay at one frequency and that at another frequency of interest in the band.

Voice vs Data

EDD has little effect on speech, however, for data transmission, it is a bottleneck for data rate.

- Undesired signals in the communication circuit
- Noise reduction is probably the most important single consideration in communication transmission
- Major limiting factor in system performance
- Noise is broken down into four categories:
	- **o** Thermal noise
	- **Intermodulation noise**
	- **o** Crosstalk
	- Impulse noise

- Occurs in all media and communication equipment, including passive devices (devices that do not produce energy, e.g. resistors)
- Occurs due to random electron motion and is characterized by a uniform distribution of energy over the frequency spectrum
- The average of this energy distribution is called white noise
- Every equipment element as well as transmission medium contributes thermal noise if the temperature of that element or medium is above absolute zero
- **•** Sets the lower limit of sensitivity of a receiving system

Directly proportional to bandwidth and temperature. Thus, the amount of thermal noise to be found in 1 Hz of bandwidth is

$$
P_n = k \times T \quad W/Hz
$$

• At room temperature,

$$
P_n = 4 \times 10^{-21} \text{ W/Hz}
$$

= -204 dB/Hz
= -174 dBm/Hz

For a band-limited system (i.e., a system with a specific bandwidth)

$$
P_n = kTB \quad W
$$

For a system with a noise bandwidth, B, measured in hertz and whose noise temperature is T we obtain

$$
P_n = -228.6 \text{ dB} + 10 \log_{10} T + 10 \log_{10} B \quad \text{W}
$$

- **Common in digital communication systems**
- Signals from two independent channels inter-modulate each other to form a product that falls into a separate band of frequencies
- Symptom: background noise
- Sources:
	- Non-linear characteristics of electronic circuits result in products of signals components.
	- Inter-modulated signals have spectral components at higher frequency bands
	- These bands may be reserved for other useful signals

- Mixing possibilities when passing F_1 and F_2 through a non-linear device
	- Second-order products: $F_1 \pm F_2$
	- Third-order products: $2F_1 \pm F_2$, $F_1 \pm 2F_2$
	- Forth-order products: $2F_1 \pm 2F_2$, $F_1 \pm 3F_2$, \cdots
- The coefficients indicate the first, second, or third harmonics
- IM noise results from either a non-linearity or a malfunction that has the effect of non-linearity

つへへ

- Unwanted coupling between signal paths.
- Three causes:
	- Electrical coupling between transmission media, such as between wire pairs
	- Poor control of frequency response (defective or poor filter design)
	- Non-linear performance in analog (FDM) multiplex systems
- There are two types of crosstalk:
	- **NEXT** is more troublesome, when transmitters and receivers are close. The signal in the Tx line is strong while it is weak in the Rx line. Twisting wire pairs reduces the NEXT
	- **FEXT** is significant when transmitting on multiple pairs in the same direction in the same time, e.g. Ethernet cables

 QQ

- Near end crosstalk is the greatest impairment
- Received crosstalk varies with
	- The volume of the disturbing talker
	- The loss from the disturbing talker to the point of crosstalk
	- The coupling loss between the two circuits under consideration
	- The loss from the point of crosstalk to the listener

- Noncontinuous, irregular pulses or noise spikes of short duration, broad spectral density, relatively high amplitude (high energy)
- **•** Symptoms: sharp sound clicks
- **Internal sources:** relay contacts, poor wrapped or poor soldered twisted pairs
- External sources: atmospheric static voltage, electric machines, microwave interferences

Voice vs Data

- **•** Impulse noise degrades telephony only marginally
- Can seriously degrade data error performance
- 1 ms impulse destroys 64 bits of 64 Kbps bit rate

Level considerations

SNR: Voice vs Data Measuring

- **Echo** is the return of a talker's voice
- It is most annoying to the talker himself/herself
- It can also be an annoyance to the listener
- It is a reflection of the voice signal caused by an impedance mismatch in the balance circuit

- **•** Excessive echo
- Singing is the sustained oscillations due to positive feedback in amplifying circuits
- Circuits that sing overload multiplex equipment particularly FDM equipment

4 **D F**

Transmission Systems Transmission Arrangement

The simple arrangement of the two wire and four wire transmission.

÷.

 298

 4 ロ } 4 \overline{m} } 4 \overline{m} } 4 \overline{m} }

- Transmission modes: Simplex, Half-Duplex and Full-Duplex
- For short distances, two-way communication is possible on a single pair of wires (bi-directional transmission).
- Problems occur however when amplification (in the past) or digital regeneration (nowadays) is needed.
- Amplifiers or regenerators in the network are uni-directional
- A Hybrid Transformer is used to convert a 2-wire circuit at the phone/terminal end to a 4-wire system in the switching network

2-Wire and 4-Wire Transmission

4 D F 一、一句 э

4 0 F ∢母 \sim

э

Hybrid Transformer

Þ \triangleright \rightarrow \equiv

41

← ロ → → ← 何 →

 299

Þ

- Hybrid **Dissipation Loss** The level of the signal (loss) in one (hybrid) 4w leg that will be transferred to the other 4w leg. It is estimated as the power, which amounts to 3dB.
- **Insertion loss**, due to an insertion of a passive element in the network is estimated at 0.5 dB
- The total loss is there fore 3.5 dB

Hybrid Transformer

- Balance network has a balance impedance of Z_N
- If $Z_N = Z_L$, then half the signal goes to the line and half goes to the balance network with little or no coupling (reflection) to the local receiver.
- But by design, we use $Z_N \neq Z_I$ to create sidetone.
	- Reflections from the CO return to the telephone set.
	- Talker hears his/her own voice.
	- Useful because acts (almost subconsciously) as a signal to the talker that the line is live.
	- No sidetone makes the line feel dead and unnatural (IP telephony often sounds like this since there's no sidetone).
- Balance Return Loss (BRL)

$$
BRL \triangleq 20 \log_{10} \left(\frac{Z_N + Z_L}{Z_N - Z_L} \right) \quad \text{dB}
$$

つへへ

Switching Systems

4 **D F**

э

 QQ

- Establishes a path between two specified terminals
- Can be done in public or private networks
- Commercial switching satisfies the following requirements
	- A user should be able to communicate with any other user
	- Connection time is not critical, but has to be relatively small compared to holding time or conversation time
	- The probability of completion of a call is also not critical but should be high (\sim 99%)
	- The user expects/assumes privacy. Usually does not specifically request it, nor, except in special cases, can it be guaranteed
	- The primary mode of communication for most users will be voice (or the voice channel)

Switching in WANs

重

Switching in PSTN

Þ \sim \sim

×.

← ロ → → ← 何 →

÷,

Switching in PSTN

Identifies subscriber Identifies local exchange

4 □

Þ

Concentration and Expansion

- Concentration is a key concept to the design of switching
- A local switching exchange concentrates traffic to reduce the number of switching paths or links within the exchange and the number of trunks connecting this local exchange to other exchanges
- A switch also performs the function of expansion to provide all subscribers served by the exchange with access to incoming trunks and local switching paths

つひひ

∢ 口 ≯ ∢ 何

 298

Þ

The telecommunication system must be very reliable. System reliability is expressed as

$$
R = \frac{\text{uptime}}{\text{uptime} + \text{downtime}}
$$

- \bullet During uptime the system is operating satisfactorily. Usually, R is expected to be 99.999%
- The system unavailability is expressed as

$$
U = 1 - R = \frac{MTTR}{MTTR + MTBF}
$$

where,

 $MTTR =$ the mean time to repair

 $MTBF =$ the mean time between failures

つひひ

Techniques to increase reliability

- **Duplicated units:** duplicated common control units, registers and processors, connected directly together and share data.
- **Standby units:** one unit is active, the rest are standby. Connected indirectly through secondary storage.
- **Load sharing:** both units are active and handle half of the calls on statistical basis.
- Spare units (Line Replacement Units): stored on-site for immediate repair of failed components.

Switching Systems Digital Switching

Þ

Main Switching Functions

÷,

Digital Switch

The heart of a modern system is a digital switch. The function of the digital switch is to provide a transparent signal path between any pair of attached devices. The path is transparent in that it appears to the attached pair of devices that there is a direct connection between them. Typically, the connection must allow full-duplex transmission.

Network Interface

The network interface element represents the functions and hardware needed to connect digital devices, such as data processing devices and digital telephones, to the network. Analog telephones can also be attached if the network interface contains the logic for converting to digital signals. Trunks to other digital switches carry TDM signals and provide the links for constructing multiple-node networks.

Control Unit

The control unit performs three general tasks. First, it establishes connections. This is generally done on demand, that is, at the request of an attached device. Second, the control unit must maintain the connection. Because the digital switch uses time division principles, this may require ongoing manipulation of the switching elements. Third, the control unit must tear down the connection, either in response to a request from one of the parties or for its own reasons.

つひひ

- Speech paths are physically separated
- Time-division switching permits a single path to be used by many calls separated in the time domain
- **•** Digital speech
- Frame concept (address!, data)
- Analogy with routers in data networks
- Voice over IP

Blocking

- May be unable to connect stations because all paths are in use
- Used on voice systems because it is expected for phone calls to be of short duration and that only a fraction of the phones will be engaged at any one time

Non-Blocking

- **•** Permits all stations to connect at once
- **•** Grants all possible connection requests as long as the called party is free
- When using data connections terminals can be continuously connected for long periods of time so nonblocking configurations are required

Space Division

 \equiv 990

∢ □ ▶ ⊣ 倒 ▶

э

PCM Switching vs Analog Switching

Advantages of PCM Switching

- Cost
	- Fewer cross-points
	- **Smaller** size
	- **Easier to achieve full availability**
- **•** Technically
	- **•** Regenerative
	- **Noise-resistant**
	- Compatibility with other digital devices
	- Lossless; no insertion loss

Switching Design

Þ

 \prec

K ロ ▶ K 倒 ▶

重

Switching Design

重

 298

÷, \rightarrow \rightarrow \equiv \rightarrow

 $\left(1\right)$

K ロ ▶ K 何 ▶

Switching Design

 \circ

It can be shown that a switch is nonblocking if

$$
k=2n-1
$$

where,

- $k =$ number of intermediate stages
- $n =$ number of inputs/outputs af each of the first/last stages

Two Functional Elements

- **Time Switch**
- **•** Space Switch

4 D F

Þ

- Since the caller and the called share time slots differently, each at their end, we need to manage this difference
- TSI involves moving the data contained in each slot from the incoming bit stream to an outgoing bit stream, with a different time-slot arrangement (storage)
- **Three basic building blocks** of the time switch
	- Memory for speech
	- Memory for control
	- Time-slot counter or processor
- Two choices for time switching
	- Sequential Input Random Output
	- Random Input Sequential Output

Time Switch $=$ Time Slot Interchanger (TSI)

- The input lines of N devices are passed through a synchronous time division multiplexer to produce a TDM stream with N slots
- To create a full-duplex connection, the incoming data in a slot must be stored until the data can be sent out on the correct channel in the next TDM frame cycle
- The TSI introduces a delay and produces output slots in the desired order. The output stream of slots is then demultiplexed and routed to the appropriate output line
- **•** Because each channel is provided a time slot in each TDM frame, whether or not it transmits data, the size of the TSI unit must be chosen for the capacity of the line, not for the actual data rate
- TSI is a simple, effective way to switch TDM data. However, the size of such a switch, in terms of the number of connections, is limited by the amount of latency that can be tolerated. The greater the number of channels, the greater the average delay tha[t e](#page-53-0)[ac](#page-55-0)[h](#page-53-0) [ch](#page-54-0)[a](#page-55-0)[n](#page-26-0)[n](#page-27-0)[e](#page-76-0)[l](#page-77-0) [e](#page-26-0)[x](#page-27-0)[p](#page-76-0)[e](#page-77-0)[ri](#page-0-0)[enc](#page-92-0)es QQ

 $Time Switch = Time Slot Interchange$

Sequential Input Random Output

4 D F

Þ

 QQ

$Time$ Switch $=$ Time Slot Interchanger

Random Input Sequential Output

4 0 F ∢母 Þ

 QQQ

- Consists of a cross-point matrix that allows the switching of time slots in the spatial domain
- \bullet The matrix consists of M input horizontals and N output verticals with a logic gate at each cross-point

重

 299

イロト イ部 トイモ トイモト

- \bullet The matrix consists of M input horizontals and N output verticals with a logic gate at each cross-point
	- If $M = N$, the switch is non-blocking
	- If $M > N$, the switch concentrates
	- If $M < N$, the switch expands
- The space array does not switch time slots as does a time switch
- Maximum capacity during any time-slot interval (simultaneous calls) $= \cdots$
- \bullet When a TSI is added \cdots

For a given time slot, the appropriate logic gate is enabled and the time slot passes from the input horizontal to the desired output vertical

€⊡

Time-Space-Time

Ξ

II

4 0 8 ← ● ÷,

Time-Space-Time

医毛囊 医牙骨下的

÷,

 299

Kロト K同下

- The space stage is considered a multiplier of the call-handling capacity
- Blocking in the overall switch occurs if there is no internal space-stage time slot during which the inlet and the outlet time-stages are idle

Space-Time-Space

4 0 8

∢●●

 298

Þ

Programmable Switching

- Previously, telecommunication switches (e.g., 5ESS) were based on large, proprietary solutions
- **•** Programmable switching is based on the open and distributed client-server computing model

c Samy S. Soliman (Cairo University) [ELCN 456](#page-0-0) Credit Hours System 66 / 93

Traditional Circuit Switching

4 D F

 QQ

- A softswitch is a general-purpose computer running specialized software
- Softswitches cost significantly less than traditional circuit switches
- A softswitch can convert a stream of digitized voice bits into packets. This opens up a number of options for transmission, including the increasingly popular voice over IP approach.
- In any telephone network switch, the most complex element is the software that controls call processing. This software performs call routing and implements call processing logic for hundreds of custom-calling features.
- Typically, this software runs on a proprietary processor that is integrated with the physical circuit-switching hardware.

Softswitch Architecture

4 D F

 QQ

- A more flexible approach is to physically separate the call processing function from the hardware switching function. In softswitch terminology, the physical switching function is performed by a **media gateway** (MG) and the call processing logic resides in a media gateway controller (MGC)
- The MG and MGC are distinct entities and may be provided by different vendors
- To facilitate interoperability, ITU-T has issued a standard for a media gateway control protocol between the MG and MGC: H.248.1 (Gateway Control Protocol)

Data Centers: Fat Tree

(2,2,4) Clos Network

4 D F

э

4 D F

Þ

K ロ ▶ K 何 ▶

 \prec э \sim -41 重

K ロ ▶ K 何 ▶

×.

É

K ロ ▶ K 御 ▶ K 君 ▶ K 君

重

K ロ ▶ K 何 ▶

×. × \mathcal{A} É

∢ ロ ▶ 《 母 》 《 ヨ 》 《 ヨ

э

 QQ

Digital Subscriber Line

4 **D F**

Þ

 QQQ

Voiceband Modem

4日下

∢母

÷,

- DSL was developed to provide higher-speed access to the Internet
- DSL is a promising technology for high-speed digital communication over the existing local loops
- DSL is a set of technologies: ADSL, VDSL, HDSL, and SDSL
- The set is often referred to as xDSL

- ADSL provides higher speed (bit rate) in the downstream direction than in the upstream direction
- Asymmetric
- The designers of ADSL specifically divided the available bandwidth of the local loop unevenly for the residential customer
- The service is not suitable for business customers who need a large bandwidth in both directions

• ADSL uses existing local loops

- ADSL reaches data rates that were not achieved with traditional modems
- Twisted-pair local loop is actually capable of handling bandwidths up to 1.1 MHz
- The filter installed at the central office where each local loop terminates limits the bandwidth to 4 kHz (sufficient for voice communication)
- If the filter is removed, the entire 1.1 MHz is available for data and voice communications
- The BW value 1.1 MHz is just the theoretical bandwidth of the local loop
- Factors such as the distance between the residence and the switching office, the size of the cable, the signaling used, etc, affect the bandwidth
- Thus, ADSL uses an adaptive technology that tests the condition and bandwidth availability of the line before settling on a data rate
- The data rate of ADSL changes based on the condition and type of the local loop cable

つひひ

- The standard modulation technique for ADSL is called the Discrete Multi Tone (DMT) technique
- It combines QAM and FDM. There is no set way that the bandwidth of a system is divided. Each system can decide on its bandwidth division
- Typically, an available bandwidth of 1.104 MHz is divided into 256 channels
- Each channel uses a bandwidth of 4.312 kHz

Bandwidth allocation:

- Channel 0: is reserved for voice communication.
- Channels 1 to 5: are not used and provide a gap between voice and data communication
- Channels 6 to 30 (25 channels): are used for upstream data transfer and control. 1 channel for control and 24 channels for data
- Channels 31 to 255 (225 channels): are used for downstream data transfer and control. 1 channel for control and 224 channels for data
- If there are 224 channels, we can achieve up to $224 \times 4000 \times 15 = 13.4$ Mbps
- However, the data rate is normally around 8 \sim 9 Mbps because some of the carriers are deleted at frequencies where the noise level is large
- In other words, some of channels may be unused

 -4 Ξ

K ロ ▶ K 何 ▶

÷,

 \sim

4 ロト 4 倒

É

4日下 ← ● **II**

 \mathcal{A}

×

÷,

- The local loop connects to a **splitter** which separates voice and data communications
- The **ADSL modem** modulates and demodulates the data, using DMT
- It also creates downstream and upstream channels
- The splitter needs to be installed at the customer's premises
- The voice line can use the existing telephone wiring in the house
- A data line needs to be installed

DSLAM: Digital Subscriber Line Access Multiplexer It packetizes the data to be sent to the Internet

4 **D F**

Þ \sim \rightarrow \prec

K ロ ▶ K 何 ▶

÷,

Thank You

Questions?

samy.soliman@cu.edu.eg

http://scholar.cu.edu.eg/samysoliman

4 **D F**