



ESCUELA SUPERIOR POLITÉCNICA DEL LITORAL
FACULTAD EN INGENIERÍA EN ELECTRICIDAD Y COMPUTACIÓN
PRIMERA EVALUACIÓN TELEFONÍA DIGITAL
SEGUNDO TÉRMINO 2018



Name: _____ Section: _____ Score: _____ /100

1. Multiple choice questions (5 points each).

a	In VoIP, voice is transmitted over which type of network? (Select 1 option)
	Circuit-based Network
	Packet-Based Network

b	According to the PTFN (Plan Técnico Fundamental de Numeración), the National (Significant) Number (N(s)N) has 9 digits and contains the following components: (Select 2 options)
	Country Code (CC)
	National Destination Code (NDC)
	Subscriber Number (SN)
	Global Subscriber Number (GSN)
	Groups of Countries (GoC)

c	What is trunk? (Select 1 option)
	Simple pair of copper wires running to your home
	Communication path between central office switch and your home
	Connects your home with central office
	Communication path between several central offices switches.

d	About ISDN... (Select 2 options)
	ISDN is used for analog voice and data transmission.
	ISDN is a standard for end to end digital transmission.
	ISDN was not globally implemented because is an ITU standard without SS7 and ISUP compatibility.
	ISDN has two types of channels: B-Channel (data) and D-Channel (signaling/control).

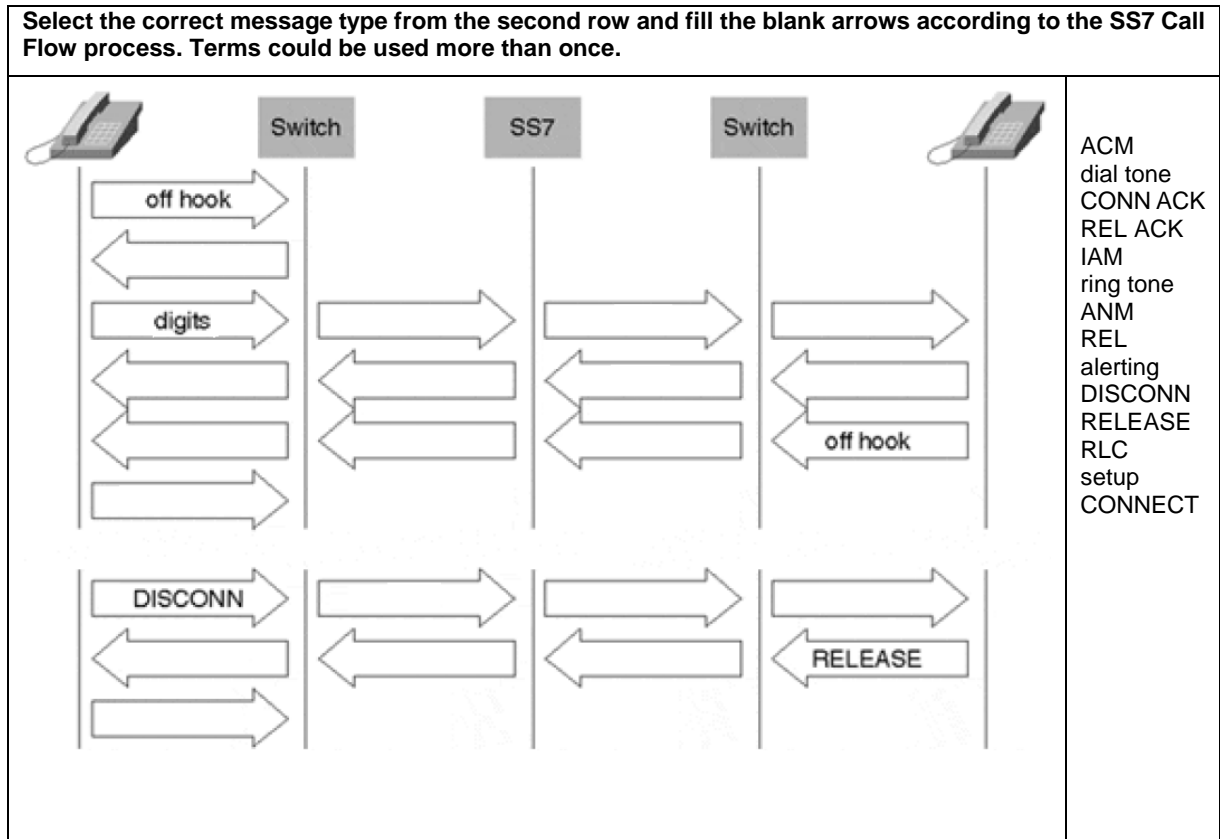
e	Which are the 4 main elements that cause degradation to voice quality in the context of VoIP?
	Jitter
	Delay
	Digital to analog conversion
	SS7 architecture
	Packet drop
	Codecs and voice compression
	Lack of bandwidth

f	Select correct statements about Voice Gateways in the VoIP architecture. (Select 3 options)
	ISDN and PSTN are used as voice compression technologies.
	Provide translation between VoIP and analog or digital medium.
	Use Digital Signal Processors (DSPs) to provide signaling and management features to the VoIP architecture.
	Use Digital Signal Processors (DSPs) to provide sampling, quantization, encoding and compression.
	Can be connected to digital or analog devices as phones, fax or other devices.

g	About codec G.711... (Select 1 option)
	Low bandwidth codec used for calls across WAN (8 kbps).
	Provides uncompressed audio normally used in LAN (64 kbps).
	Optimized for calls across the WAN (13.3 kbps).
	Wideband codec which provides high audio quality for 8900/9900 phones (10 kbps-32 kbps)

h	Which are recommended parameters to provide good voice quality? (Select 3 options)
	Traffic loss (tail drop) less than 90%.
	End-to-end delay less than 150 ms.
	Variable delay less than 5000 ms.
	Traffic loss (tail drop) less than 1%.
	Variable delay less than 30 ms.
	Using only G.711 codec.
	Using only G.729 codec.

2. Complete the diagram question (10 points).



3. Essay questions (10 points each).

- (a) Define the advantages of digital signaling over analog signaling in telephony (also, specify analog signaling main problem).**
- (b) Describe the PCM process for telephony.**

4. Diagram + essay question

- (a) Using a diagram to plot the SS7 Network Architecture (must contain all the signaling links and elements). (20 points)**
- (b) Describe each signaling element of the SS7 Network Architecture. (10 points)**