PRACTICAL WORKBOOK

FOR ACADEMIC SESSION 2011

TELECOMMUNICATIONS SWITCHING SYSTEMS

(TC-485)

FOR

BE (TC)

Name: ______________________________

Roll Number: ______________________________

Batch: ______________________________

Department: ______________________________

Year: ______________________________

Department of Electronic Engineering
NED University of Engineering and Technology, Karachi

LABORATORY WORK BOOK

For The Course

TC-485 Telecommunications Switching Systems

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LAB SESSION #1

LOCAL SIGNALING AND LINE SCAN

OBJECTIVES:

To become familiar with:

- Operation of a Single Digital Switching Centre
- The call protocols with respect to originating and called terminals
- To determine the acquisition procedure for on/off hook and DTMF
- To investigate the concept of scanning line circuits to receive local signals
- To evaluate the associated minimum sampling rates

THEORY 1(a):

Local Signaling:

The Lab is largely concerned with signaling between the telephone user, and the local Switching Centre. This is known as local signaling. The signals available to the user are the **Switch Hook** and the **Keypad**. The Switch Hook operates as soon as the telephone is lifted. This is the Off Hook state of the telephone, and is recognized by the Switching Centre.

The Keypad is primarily used to send the Destination Address to the Switching Centre; that is the number of the telephone to which the connection is required. The signals are in the form of a combination of two audible tones, a different combination for each number on the Keypad. Hence it is known as **Dual Tone Multi-Frequency (DTMF)** signaling. The Switching Centre can send signals to the user, by using *tones* and by *ringing* the bell or alerter in the telephone. The audible tones are known as call progress tones, and indicate to the user important responses of the Switching Centre. Obviously they are only useful if the user is listening to the telephone. If the telephone is not in use, i.e. if it is on hook, then the Switching Centre can *ring* it.

ITU-T Standards:

Standards for the telephone industry are agreed by an international body. The Standards used in these Labs were all issued by the CCITT before 1994, but have been adopted as ITU-T Standards. The ITU-T produced a Standard Recommendation E. 180 for the tones used in local signaling.

Each telephone system is run by an Administration, usually running the whole telephone system in one country. Historically, each Administration has often used different tones for the same purpose. The ITU-T Recommendation aims to reduce these differences so that in international calls operators and users understand easily the meaning of the tones. The Recommendation includes 'acceptable' tones for each purpose, and 'recommended' tones for new systems. The general nature of each tone and the acceptable and recommended limits are:

Dial Tone should be a continuous tone. Either a single frequency in the range 400-450 Hz, with 425 Hz preferred, or a combined tone of up to 3 frequencies, with at least one frequency in the ranges 340-425 and 400-450 Hz, with at least 25 Hz difference between any 2 frequencies. However, any existing dial tones, including interrupted tones are acceptable, because of the technical and social difficulties of changes.

Ring Tone is a slow period tone, in which the tone period is shorter than the silent period. The recommended limits are 0.67 to 1.5 seconds for the tone and 3 to 5 seconds for the silence; and the acceptable limits are up to 2.5 sec and up to 6 seconds respectively. The recommended frequency is 400-450 Hz, with 425 Hz preferred; and the acceptable range is 340-500 Hz.

Busy Tone is a quick period tone, with the tone and silence periods equal. The total duration of both tones is recommended to be 0.3 to 1.1 seconds; and the ratio of tone to silent period should be between 0.67 and 1.5. The recommended frequency is 400-450 Hz, with 425 Hz preferred; and the acceptable range is 340-500 Hz. Number Unobtainable Tone, no recommendations are made.

Operation:

The normal use of the 4 telephones connected to the local Switching Centre is demonstrated in this Practical. By using the telephones to make calls, the basic operation of the Switching Centre is examined.

For the first few lab sessions, including this one, the telephones use single digit numbers. The numbers to be dialed correspond to the Line numbers L1 to L4 shown on the Work board; i.e. 1, 2, 3 and 4.

The System uses 4 Call Progress Tones: Dial tone, Ring tone, busy tone and Number Unobtainable (NU tone). Using the telephones the tones can be heard. Also the connection of them can be seen on the Switching Centre diagram. There is both a preset set of tones, and a programmed set.

Programming the tones is performed in the Tones and Cadences Practical.

Ringing the telephone requires a much larger voltage than the acoustic Tones. This is indicated by a different colour inside the Switching Centre diagram. The Digital Switching Centre Work boards used for this can operate as one of two types, **A** or **B.** The Work board used for the Single Switch Assignments must be set to type **A,** by the ***Switching Centre Type*** switch at the far right hand corner of the board. If two boards are connected, the other should be type B, and is not used for these Assignments.

PROCEDURE 1(a):

- This exercise is to become familiar with the use of the 4 telephones connected to the local switching centre. The numbers for the telephones for the first session correspond to the line numbers on the work board; i.e. 1 to 4. The work board must be set to Type **A.** If there is another board connected, it must be type **B,** and is not used. The tones are preset but non-standard. They can be set to familiar tones in the Tones and Cadences practical.

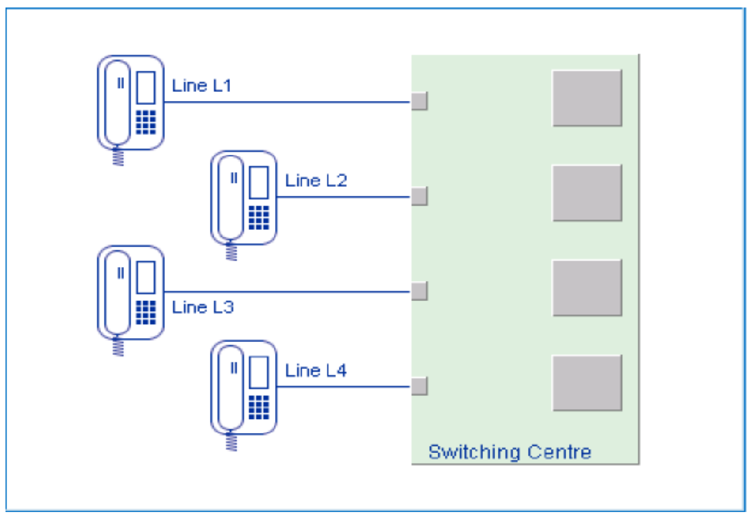

Figure 1.1 Basic diagram of a Switching Center with telephones connected

- Put all 4 telephones face down (On Hook).
- Pick up telephone 1. Describe the signal heard.

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- Press button 2. Note down the changes experienced.

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- Pick up telephone 2. Speak into one of them to check the connection.

- While the first connection is held, pick up telephone 3 and press button 1. What do you hear?

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- Replace telephone 2. Is the connection broken? ____________________. Replace telephone 1.

- Using any telephone, listen to Dial Tone, and then press button 8. Write your observation.

____

- Try out other connections.

QUESTIONS:

1. If a call is made from telephone 1 to telephone 2, what happens if telephone 2 is replaced (Switch Hook pressed), and then picked up again?
2. With the same call what happens if telephone 1 is replaced?
3. What tones or messages do you expect for Dial Tone, Ring Tone, Busy Tone and Number Unobtainable, when using your normal telephone?

THEORY 1(b):

Line Scan:

There are four telephones connected to each Switching Centre Work board. Each is connected to a Subscribers Line Interface Circuit (SLIC), which provides the services demonstrated on the two Telephony Work boards Telephone and Interface and TDM/PCM Principles.

The SLIC circuits are on 20-pin single in-line packages, which are visible just behind the telephone sockets on the Switching Centre Work board.

The 2 outgoing signaling circuits in each telephone are the Switch Hook and the Keypad

Switch Hook Detection:

A Subscriber's Line Interface Circuit (SLIC) detects the state of the Switch Hook in each telephone. There are *3 conditions* in which the state of the Switch Hook can be determined.

1. If the telephone is On Hook, and not in use, or

2. If two telephones are connected; then a Call Detect circuit determines the state of the Switch Hook.

3. However, if the telephone is receiving a Ringing signal, then the normal speech circuit is disconnected and an Answer Detect circuit is used to perform the Ring Trip function.

The SLIC used on the Work board performs both functions, and only produces one Switch Hook signal. However, it may be possible to detect a difference in response time when the Ringing is audible.

DTMF Detection:

DTMF Receiver circuits recognize the output from each keypad.

The control processor obtains information about the all the lines by scanning the Interface Circuits to receive any input signals.

Each valid detection is signaled by an output pin going positive for at least 40 ms. To ensure that no valid inputs are missed the scan is repeated every 20 ms. That is 50 times every second.

Switch Hook:

The control microprocessor in the Switching Centre needs to know the state of the switch hook in each telephone. Is it On Hook or Off Hook? The Subscriber’s Line Interface Circuit (SLIC) on each Line, tests the state of the Switch Hook continuously. The processor scans all 4 SLICs in one 4 bit input.

The scan is repeated at a standard 20 ms interval. The processor only needs to respond if there has been a change of state. For example, if a telephone was On Hook previously and is still On Hook, then no action is expected.

Ring Trip:

The SLIC normally uses a Call Detect Circuit to test the Switch Hook. However, if the telephone is receiving a Ringing signal then an Answer Detect circuit is used. That function is known as ring trip. Ring trip takes a little time to operate, as the change in dc current has to be recognized in the presence of a large ac current. A typical response time for Ring Trip is 200 ms, which may be just detectable.

PROCEDURE 1(b):

- Each telephone may go Off Hook while either idle or while being alerted. One Switch Hook signal is operated in either case.

Switch Hook State				
Line	L4	L3	L2	L1
Bit	3	2	1	0
Switch Hook	0	0	0	0

Figure 1.2. Switch Hook State

- Pick up any telephone and observe the change of state.
- Replace the telephone and see the response.
- Use one telephone to alert another. (Single digit dialing, 1 to 4). Pick up the telephone which is ringing. Replace the calling telephone. Record your observations.

QUESTIONS:

1. Can you detect a difference in speed of response to the Switch Hook when the telephone is starting a call (Call Detect), or answering in the silent period, or when it is ringing (Answer Detect)?

2. Why is there a difference?

3. Consider a call which has been established from Line L3 to Line L4. What action is expected by the control if:

    - Bit 3 changes from 1 to 0.

      ____

    - Bit 2 changes from 1 to 0.

      ____

THEORY 1(c):

DTMF Receivers:

The keypad sends Dual Tone Multi-Frequency (DTMF) signals to the Switching Centre. Each line circuit has its own DTMF receiver circuit in a dedicated IC. The DTMF receiver responds only to valid DTMF tones. When it recognizes a valid tone, it sets an output pin to a binary 1. This output is interpreted by the microprocessor as a DTMF Valid (DV) signal. The DTMF receiver only holds the DV output for about 40 ms. It cannot be longer because another DTMF tone may be transmitted within about 80 ms.

The processor scans all 4 DV outputs at once in its regular line scan. The repetition rate of the scan is set at 20 ms, in order to ensure the reliable response to the DTMF tones. The 4 bit DTMF codes are appear continuously on the output pins of the DTMF receiver, but of course are only significant when the DV signal has appeared. The DTMF Receivers are connected permanently, so that they can be used during a telephone conversation. Their response can be seen at any stage of a call.

It is necessary to input one digit for each time a button is pressed. Thus if DV stays at 1, without interruption, it is still only one digit. Hence the control only responds to a positive change, that is when the DV signal changes from 0 to 1. After the button is released, the DV returns to 0.

PROCEDURE 1(c):

- Each line has its own DTMF Receiver for tone dialing.
- When a receiver has recognized a valid tone, the DV (DTMF Valid) bit is set; and the digit can

be read from the 4 bit code.

DTMF Valid - DV				
Line Bit	L4 3	L3 2	L2 1	L1 0
Input	0	0	0	0

DTMF digits				
Bit	3	2	1	0
Line L1	0	0	0	0
Line L2	0	0	0	0
Line L3	0	0	0	0
Line L4	0	0	0	0

Figure 1.3. DTMF valid status of telephone lines

- Use the keypad on different telephones. Observe and record the response of the DV and digit codes.

QUESTIONS:

1.Under what condition should the microprocessor control use the DTMF code for any particular line?

2.Why does the Line Scan occur every 20 ms?

3. How does the control determine whether a keypad button has been pressed more than once?

LAB SESSION # 02

CALL RECORDS

OBJECTIVES:

- To recognize the use of calling and required line identities and establish the means of storing the associated information
- To learn about the stored information needed to control each telephone call
- To establish methods of timing call processes to allow efficient system utilization

THEORY 2(a):

Call Records:

A Call Record is an area of memory in which the essential data for any call is held. Each time a telephone goes off hook in order to start a call; a new Call Record is opened.

The essential items are:

1. The *call state,* which indicates the position which a call has reached in the sequence of events.
2. The identity of the *calling line.*
3. The identity of the *required line*
4. The *duration* of the call.
5. The *timing* of the Ring Count.

These items, together with the Line Scan inputs, are the total requirements for the Control microprocessor to handle the call.

The Call Records are numbered R1 to R4. For a very small system it is possible for all telephones to be Off Hook together and so there has to be a Call Record for each one. However, you will probably notice as you work with the system that it is unusual for all 4 to be needed.

In large systems an assumption is made that most telephones are not in use most of the time, and so only a small proportion of possible connections are ever made.

Line Identity:

For the Call Record the lines are identified by their Equipment Numbers, which correspond to the Telephone Line numbers L1 to L4 on the Work board. The use of alternative numbers is discussed in the Line Records.

There are 2 Line Identities used in the Call Record. The Calling Line Identity *CLI* is entered when the Call Record is opened. The Required Line Identity *RLI* is entered when it is identified by the dialed numbers.

Call Progress:

The basic information required by the Control to process a call comprises the Call State CS and the Line Identities CLI and RLI.

1. With that information, incoming signals from the telephone are correctly interpreted.
2. All switching whether of Tones or Ringing or final Connection can be performed.
3. Release of a call may happen at any State of the call. Hence correct disconnection also requires all the 3 items, CS, CLI and RLI.

PROCEDURE 2(a):

- There are 2 Line Identities required for the Call Record.
- The ***Calling Line Identity CLI*** is entered into the Call Record when a new Record is started.
- The ***Dialed Number*** is stored as the digits are dialed. When it is complete, if the required line is not Busy or Unobtainable, the ***Required Line Identity RLI*** is entered into the Call Record.

Call Record	R1	R2	R3	R4
Calling Line Identity CLI	-	-	-	-
Dialled Number	-	-	-	-
Required Line Identity RLI	-	-	-	-

Figure 2.1. Call Record Table

- Use the telephones to make various connections (2 digit dialing). Observe and note when the Identities are entered and removed from the Call Records.

QUESTIONS:

1. When is a new Call Record started and when is it cleared?

2. If a Call is made from Line L1 to Line L2, at what stage of the Call is the Required Line Identity entered into the Call Record? What is the Call State when it appears?

3.Consider a Call at State S4 with CLI = L2 and RLI = L3. What is happening?

Now consider the following signals:

(i) Line L3 goes Off Hook.

(ii) Line L1 goes Off Hook.

(iii) Line L2 goes On Hook.

(a) For each case what Switching is required?

(b)For each case what changes are made to the Call Record?

4. What is the essential information required in the Call Record for the control of a call?

THEORY 2(b):

Timing:

Call Duration CD:

The duration of any telephone call is important to a telephone company for various reasons. The main reason is for charging the user for the call. When the call is completed, the details of the call are recorded and used for calculating the bill. Also it is useful to keep some statistics on the system performance. For example, the time required to set up a call, for which the company is not paid, affects the amount of computing equipment needed by the system.

Finally, if calls are not completely established, e.g., a phone is left off-hook without dialing, after a certain period it may be disconnected or an alarm message may be sent.

Ring Count

According to the ITU-T recommendation, Ring Tone and the Ringing signal should commence as soon as the connection is made. To achieve this without distorting the normal Ring cadence, the Tone period must start immediately. Hence the cadences for each line are not synchronous, and must be counted independently.

PROCEDURE 2(b):

- The duration of each call, in seconds, is maintained in each Call Record. Timing starts as soon as the Call Record is opened.
- When a connection is made, the Set Up duration is recorded for the Traffic statistics. The Call Duration is then restarted.
- The control of Ringing also needs a location for counting the course of the Ringing cadence. The Ring Count is counted in units of 0.1 second.

Call Record	R1	R2	R3	R4
Call Duration CD sec	0	0	0	0
Ring Count	-	-	-	-

Figure 2.2 Call Record Status

- Make some connections (2 digit dialing), and observe the timing.

QUESTIONS:

1. Why is the duration of a call measured?

2. Why is the setup time measured?

3. Would a telephone company welcome automatic attempts to make a connection to a busy line?

4. Does your local telephone company disconnect calls if dialing is not completed? How long do they wait?

LAB SESSION # 03

DIGITAL SWITCHING PRINCIPLES

OBJECTIVES:

- To establish the digital signal path in time and space, between two terminals
- To evaluate the time and space parameters of a switched signal
- To investigate the organization of a multi-bus switched system

THEORY 3(a):

Digital Transmission:

Digital telephone signals use a **synchronous** transmission system, which combines Time Division Multiplexing (TDM) and Pulse Code Modulation (PCM). Each speech path has to send an 8 bit binary code at a rate of 8000 codes per second.

The codes are organized into groups called **frames**. Each code is transmitted in a timeslot. The frames include special synchronizing codes so that each timeslot can be identified and the code correctly converted back to analogue form.

In the **CEPT system** which originated in Europe there are 32 timeslots in each frame. Each frame takes 125 µs, and each 8 bit timeslot is transmitted in 3.9 µs, at a rate of 2.048 Mbps. In the **T1 system** which was designed in North America there are 24 timeslots in each frame. Each frame also takes 125 µs, and so each 8 bit timeslot is transmitted in 5.2 µs. An extra bit is used for synchronizing so the transmission rate is 1.544 Mbps.

The connection to each telephone uses 2 wires which carry analogue signals in both directions. For Digital Switching, incoming and outgoing speech are separated by **hybrid circuits.** Then combined **Codec/Filter** circuits provide analogue to digital and digital to analogue conversion.

Speech Paths:

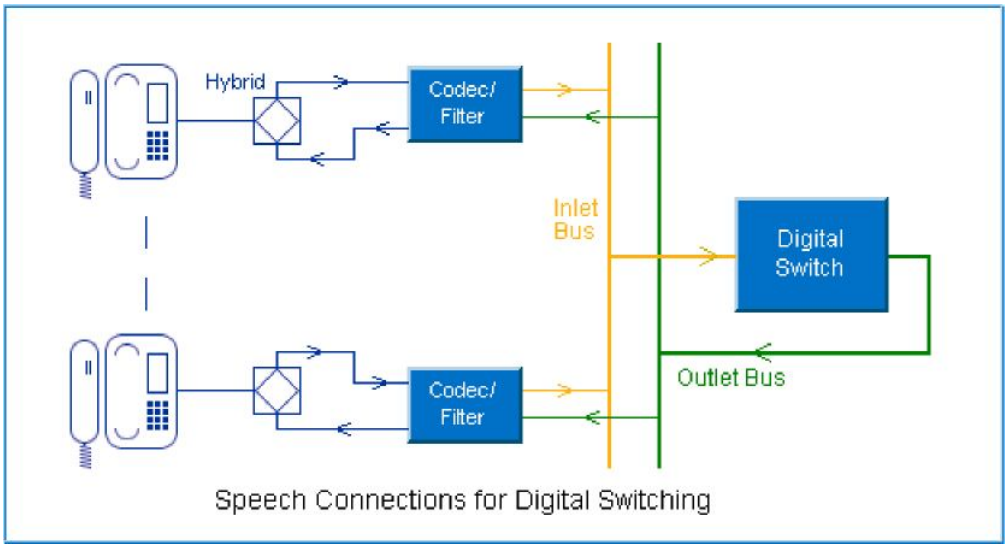

Figure 3.1. Speech Circuit Connections for Digital Switching

The Codec/Filter circuits are connected to the Digital switch through Inlet and Outlet Buses. Each Bus can carry up to 24 or 30 speech connections, depending on the PCM system in use. Each Codec is connected to the Inlet and Outlet Buses at a designated time; for the period of one timeslot.

Basic Digital Switching:

The Digital Switch transfers the contents of each timeslot in the Inlet Bus to the appropriate timeslot in the Outlet Bus. Switching of the data from each timeslot in the Inlet Bus requires changing the time at which the data is transmitted along the Outlet Bus. The process is known as Time Switching or Time Slot Interchange

Time and Space Switching:

Most switching systems have more than 24 or 30 channels and use more timeslots than can be accommodated in one Bus. Therefore switching between the Buses is required.

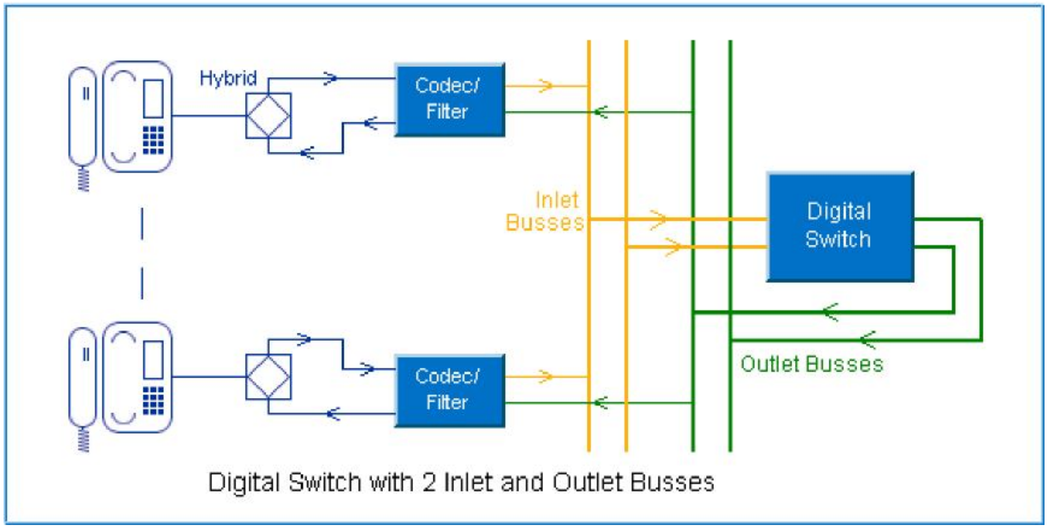

Figure 3.2. Inlet and Outlet Buses in the Digital Switch

The Digital Switch shown must switch between different timeslots and also between different busses.

Time Switching:

The basic process in Digital Switching for telephone systems is the transfer of 8 bits of digital data from one Timeslot to another. The speech signals from each telephone are connected through a Codec to the Switch during a specified Timeslot. The connections are made along Inlet and Outlet serial buses. The simplest connection between 2 telephones occurs if they are both using the same Inlet and Outlet buses. They must, of course, use different Timeslots.

Then 8000 times per second the contents of the Inlet Timeslots for each telephone must be transferred to the Outlet Timeslots of the other. This is **Timeslot Interchange or Time Switching**. The Timeslots are organized in Frames. Each Frame has 32 (CEPT systems) or 24 (T1 systems) Timeslots. Successive Frames are transmitted along the same physical connections i.e. the same Inlet and Outlet buses.

The data for transmission is only available briefly, and the display flashes to suggest this. Of course the actual data transmission is very much faster than the flashing. For convenience, one digit dialing is still used in this Lab (Line numbers 1 to 4).

PROCEDURE 3(a):

- Switching between 4 telephones, all of them connected to one bus. The data is only available briefly in each timeslot, as suggested by the flickering display.

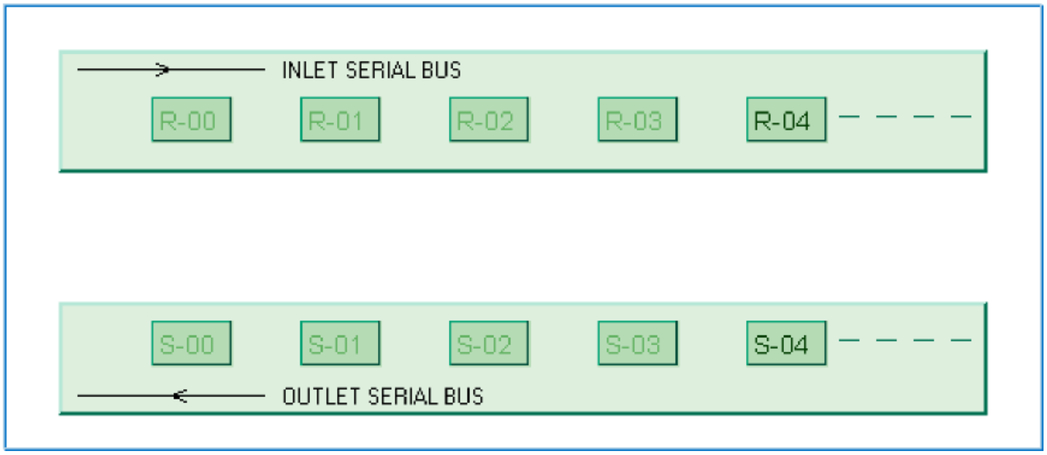

Figure 3.3 Data Traffic in the Inlet and Outlet buses

- Make a connection between any two telephones. They have numbers 1 to 4. Observe the timeslots which have their contents exchanged to carry the speech signal from one telephone to the other.

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- Make another connection without breaking the first one and observe the new timeslot interchange.

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- Clear the connections and make new connections.

QUESTIONS:

1. What is the essential process in digital switching?

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2. How many data transfers between timeslots are required for one telephone connection?

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3. How often are the data transfers made?

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4. How long is each data sample available on the serial bus?

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5. How many connections can be made between telephones on one 30 channel bus?

THEORY 3(b):

Time & Space Switching:

One 30 channel CEPT serial bus can accommodate 30 telephones, and each T1 serial bus can accommodate 24 telephones. Hence any larger system requires more Inlet and Outlet serial buses, particularly if they are public systems. Hence switching between serial buses is required as well as time switching. This is known as combined **Time and Space Switching**. This lab involves two Inlet and two Outlet serial buses.

In a large system there are many such serial buses, and complex combinations of Time and Space switching are used.

PROCEDURE 3(b):

- The 4 telephones are now connected to 2 timeslots in different buses.

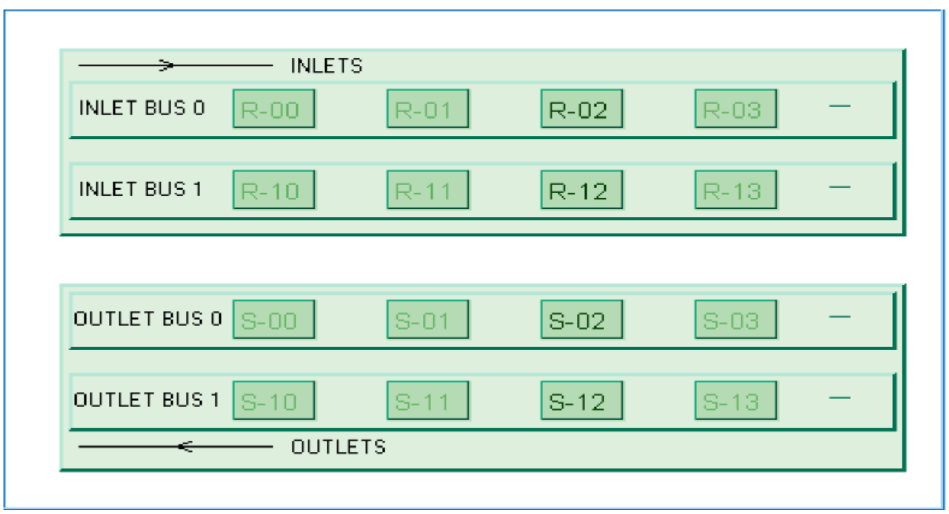

Figure 3.4. Timeslots in Inlet and Outlet Buses

- Again make different connections between the telephones. They are numbered 1 to 4.

- In each case observe the timeslot interchanges required. Some of them are in the corresponding serial bus, for example from Inlet bus 0 to Outlet bus 0 but some require speech data to be exchanged into an unrelated bus.

QUESTIONS:

1. Why is space switching sometimes required as well as time switching?

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2. How many telephones can be connected through a digital switch with 8 Inlet and Outlet 30 channel serial busses?

3.Consider how several similar switches could be connected to provide for 16 inlets and outlets?

LAB SESSION # 04

DIGITAL SWITCH OPERATION

OBJECTIVES:

- To learn how the Data Memory can produce Digital Switching, and how it is controlled by the Connection Memory.
- To determine the relationship between information held within the control memory and the actual switching operation.
- To establish the procedure to connect tones to calling lines.

THEORY 4(a):

Digital Switch Architecture:

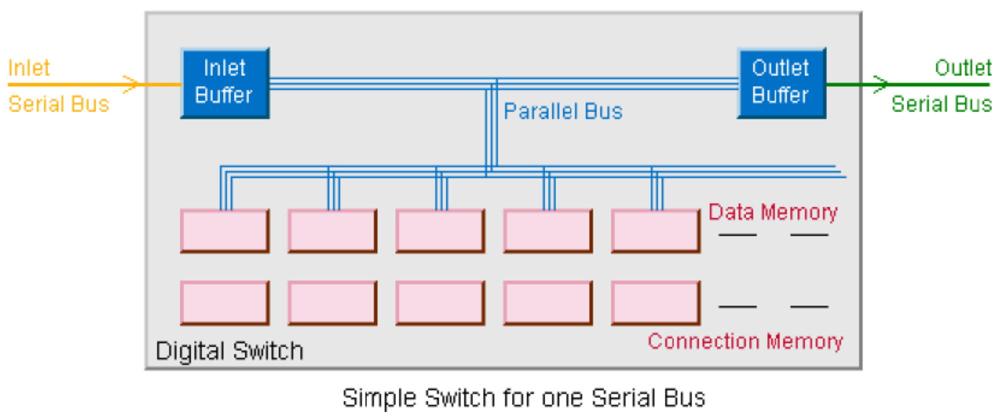

Figure 4.1. Digital Switch Architecture

The Switch has an Inlet Buffer (Serial In and Parallel Out), and an Outlet Buffer (Parallel In and Serial Out) for each Serial Bus connected. The Buffers and the Data Memory are connected internally by a parallel bus, which can operate much faster than the serial busses. The Connection Memory is used to control the time at which the contents of each timeslot are sent to the Outlet Buffer.

Switch Operation:

The incoming serial data along one Inlet Bus enters the Inlet Buffer. The data in each timeslot is read into a location in the Data Memory in sequence as it arrives. Thus the position in the Data Memory indicates which timeslot was connected. To achieve Time Switching, the data from each location in the Data Memory must be read at the correct time, and sent to the Outlet Buffer.

Each location in the Connection Memory is used for the same timeslot as the corresponding location in the Data Memory. However each Connection Memory location contains an address, not data. The address is that of the Data Memory location which should be read and the data sent to the Outlet Buffer for each timeslot. Thus by writing data into the Data Memory sequentially, but reading it when required, Time Switching is obtained. Combined Time and Space switching is obtained using the same principle. More Inlet and Outlet Buffers are used for the additional busses, all connected by the parallel bus to the Data Memory, which is more extensive. A limit on switch capacity is reached when the internal bus cannot reach the speed required for transfer of all the data. In that case combinations of switches are used.

Control of Time Switch:

Switching using RAM:

The method used for Time Switching is to write the 8 bits of data in each Timeslot into a location in a random access memory (RAM). Then at the correct time the data is read from the RAM and transmitted through the Switch Outlet. To do this the Digital Switch writes the data from each Timeslot in the Inlet serial Bus into the Data memory. Each location in the Data Memory corresponds to a particular Timeslot in the Inlet Serial Bus. The data is read out from the Data Memory and sent to the Outlet Serial Bus in the correct sequence for the particular connection required.

Switch Control:

The output sequence is controlled by the Connection Memory. Each location in the Connection Memory also corresponds to a Timeslot in the Outlet Serial Bus. The Connection Memory contains the *addresses* of the Data Memory from which each Timeslot data can be read. The addresses are inserted when the connection is set up.

At the time for the data to be transferred to the Outlet, the address is read in the corresponding location in the Connection Memory, and used to find the data in the Data Memory. The data is held for a complete Frame; i.e., 125 μs, in the Data Memory before being written over by the data from the next corresponding Timeslot. This is indicated by the display not flashing as quickly as the serial data.

PROCEDURE 4(a):

- Control of connections between 4 telephones, all connected to one serial bus. Make a connection between any two telephones. (Single digit dialing, numbers 1 to 4)

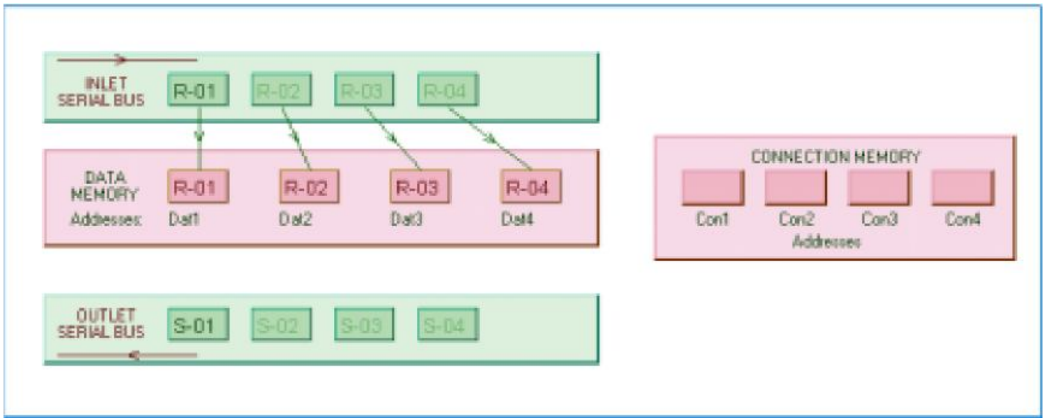

Figure 4.2. Connection memory and Data Memory

- Observe the addresses written into the ***Connection Memory.*** What do the addresses mean?

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QUESTIONS:

1. How many locations in the Connection Memory need to be written for one speech connection?

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2. How long does each sample of data remain in the Data Memory?

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3. What is the sequence of actions to find the correct data for the Outlet during a particular timeslot?

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4. What entries are made in which locations in the Connection Memory for a connection between lines L2 and L4?

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THEORY 4(b):

Connection of Tones:

The Digital Switch is used to connect each of the 4 tones used for signaling to the telephone user. The tones are Dial tone (DT), Ringing tone (RT), Busy tone, and Number Unobtainable (NU tone).

The tones are continuously available in particular timeslots, and are connected to each line as required. Each tone can be connected to as many telephones as necessary simultaneously. Whether it is the programmed or preset tones that are used, they are available in the same timeslots.

PROCEDURE 4(b):

- The 4 tones for signaling to the telephone are available in specified timeslots, in one serial bus. They are connected as required to the telephone timeslots, again under the control of the connection memory.

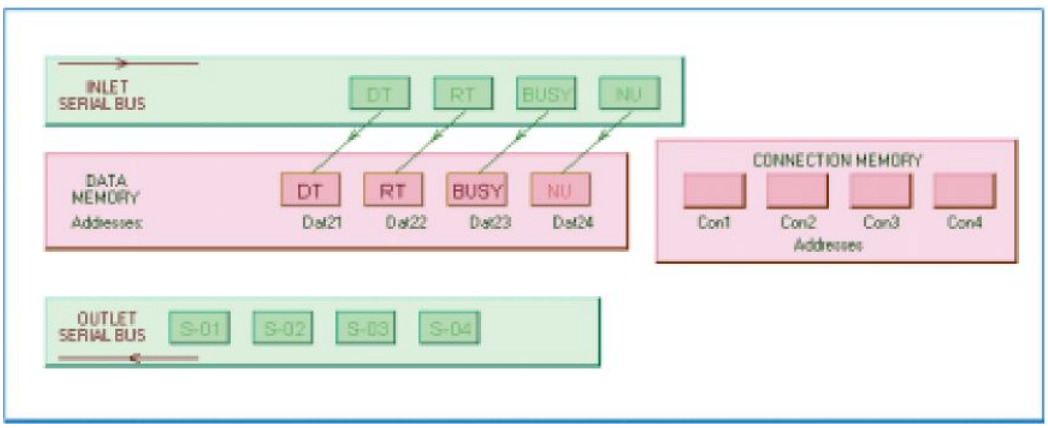

Figure 4.3. Data Memory and Connection Memory

- Pick up one telephone and listen to the dial tone. Dial another telephone, see that dial tone stops, and then ring tone starts. Answer the telephone.
- Then try to connect a third telephone to one of the first two. Finally try to connect to an invalid number.

QUESTIONS:

1. What address is entered into and in which location of the Connection Memory, in order to apply Dial tone to Line L3? Why?

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2. Is there any limit to the number of lines which can receive the same tone from one source at the same time?

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LAB SESSION# 05

LINE RECORDS AND LINE MAPS

OBJECTIVES:

- To establish the relationship between directory and equipment numbers.
- To analyze the association between the location in space, and the time slot used, for a particular channel.
- To examine the Records relevant to each telephone line
- To evaluate the call accounting procedure.
- To analyze the use of the system memory to determine whether a required line is free or in use.
- To determine the procedure used to apply ringing to a required line.

THEORY 5(a):

Line Records:

This Lab session examines the Line Records which are related to each line into the switching centre. Every Line has to have one. They are permanent records, unlike the Call Records which are closed as soon as a Call is completed. However, the degree of permanence is different, depending on the nature of the Record.

A Directory Number is allocated to a Line when it is installed, and it would be unusual to change the number when in service. Call Accounting Records are added to whenever a Call is completed. They are kept as long as necessary for charging, but can eventually be discarded.

Numbering:

Line Identity:

Each Line is identified in different ways in the system for different purposes.

1. The Equipment Number for each telephone is the Line Numbers L1 to L4 marked on the Work board.

2. The Digital Identity is derived from the physical connections on the Work board, and the

timing arrangements. Each Line is connected through the Codec to a particular serial bus in the Digital Switch. A timing pulse or a digital control defines a timeslot on that serial bus.

That combination of Serial Bus and Timeslot provides the Digital Identity for the Line and is used by the software for control of the switching.

The switch has 8 Serial Bus Inputs. Thus 3 binary bits are required to define each one. There are 32 timeslots in each Serial Bus, to accommodate the CEPT system, and 5 bits are required for this. Thus a total of 8 bits are required. The Serial Bus is represented by the most significant bits.

The Digital identity can be expressed in binary or in hex form. The Digital identity as well as the Equipment Number is permanent.

3. The Directory Number is the number that is dialed by the user.

Unlike the other identities, the Directory Numbers are held in software, and can therefore be changed. A Directory Number is also known as a Destination Address.

PROCEDURE 5(a):

- There are 3 different methods of identifying each Line, the Line number, a Digital Identity, derived from the Digital Switch address and the Directory Number.
- A Directory Number can be easily changed by software.

Line	Digital Identity									Directory Number
	Binary								Hex	
	S	S	S	T	T	T	T	T		
L1	0	0	0	0	0	0	0	1	01	21
L2	0	0	0	0	0	0	1	0	02	22
L3	0	0	1	0	0	0	0	1	21	31
L4	0	0	1	0	0	0	1	0	22	32

Change Directory Number

Figure 5.1. Changing the Directory number

- By using the ***Change Directory Number*** button, the Directory Number for any line can be changed to any 2 digit number. Change a Directory Number and use it to make a connection.

QUESTIONS:

1. Why are 5 bits allocated to recording the Timeslot identity of each line?

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2. What are the binary and hex Digital Identities for Line L3?

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3. What would the Digital Identity be for a Line connected to timeslot 14 on serial bus 5?

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4. When Directory Numbers are usually allocated to a particular Line?

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LINE ACCOUNTING:

PROCEDURE 5(b):

Call Accounting Record for Line L1				
Item	Call Duration sec	Destination	Time Hr:Min	Date
0	0	0		0
1	0	0		0
2	0	0		0
3	0	0		0
4	0	0		0

Select Record

Figure 5.2. Call Accounting Record for Line 1

- Full details of each Call are recorded as they are finished. In a commercial system they would be used for itemized billing and charging.
- The date and time are derived from your computer. If the computer is not set properly, they will be incorrect.

Record your observations:

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- The Call Accounting Records for each Line are found with the ***Select Record*** button.
- Use the telephones, and observe the entries in the relevant Records. The last 5 calls made on each line are available.

QUESTIONS:

1. Why is it necessary to record the destination, the date and time of each Call?

____

2.Why is the duration of each Call recorded? ____

3.Are there any Calls in your local systems which are free? ______________________________

4. If so, how can they be identified in the Call Accounting Record? _______________________

Line Maps:

THEORY 5(C):

The Line Maps are systems used by the Switch Control to simplify and speed up operation. The switch control continuously monitors the input from the Line Scan. When a change is detected the control updates the relevant Call Record, and then takes the appropriate action. To achieve this promptly it needs to find the existing state of the line quickly.

Line Maps contain information about the state of the system, arranged in the sequence of the external Lines. They act as a cross-reference between the Line Scan and the Call Record and contain different amounts of information about each Line. The *Location Map* has the identity of the Call Record, if any, which refers to that Line. The *Condition Maps* only have one binary bit per Line. The *Busy Line Map* provides a rapid check for whether a required line is busy or not. The *Ring Output Map* helps in control of the ringing circuits.

Location Map:

During the Line Scan, a signal may be received from a particular Line. The significance of the signal depends on the State of the Call. The State is recorded in a Call Record. However with a large system it would be a time consuming operation to search through all the Call Records to find the Line identity.

The Location Map shows which Call Record, if any, has a reference to each Line, so that the relevant Call Record can be found immediately.

Operation:

If a Line goes Off Hook, and there is no entry in the Location Map for that Line, then it is treated as a new Call, and a new Call Record is opened. The Call Record number is inserted at the Calling Line Identity CLI address. As the Call is being set up, when the Required Line Identity RLI is identified, it is entered into a Call Record if it is free. The Call Record number is entered into the Location Map at the RLI address.

Thus any entry in the Location Map shows that a Line is busy. When the RLI goes Off Hook, the Control examines the Location Map and finding an entry, updates the Call Record and takes the appropriate action.

Procedure 5(c):

- This Map shows which Call Record contains a reference to a particular Line. It provides the first location where that Line is already referred to.

Location Map				
Line	L4	L3	L2	L1
Call Record	-	-	-	-

Figure 5.3. Location Map

- Make a connection between any two telephones. Observe the entries written into the map.
- Make more connections, or simply lift the telephones off-hook, and observe the changes to the map.

QUESTIONS:

1. If Line L1 goes Off Hook, and there is no entry in the Location Map for that Line, what State has the Call reached? What changes should be made in the Call Record?

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2. If Line L2 goes Off Hook, and there is an entry in the Location Map for that Line, what state has the Call reached? What changes should be made in the Call Record?

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3. If a Line goes On Hook, must the Control find an entry in the Location Map?

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LAB SESSION # 06

TESTING AND TRAFFIC

OBJECTIVES:

- To evaluate the inherent facilities within the system to test some of the SLIC, CODEC and digital switch functions.
- To investigate the traffic capacity of the system.
- To evaluate methods of collecting traffic data.
- Understand the means of measuring traffic.
- Understand the implication of the traffic level on system design.

THEORY 6(a):

Testing:

Telephone systems in operation need to be **tested** regularly to ensure correct operation. Tests usually consist of the transmission of a test signal over part of a system, and its detection. The detection was originally performed by simply listening to the signal. Now, as far as possible, it is carried out automatically. Testing is generally performed at times of low usage, for example during the night. It is also necessary to ensure that any equipment being tested is not required for use. The practicals introduce some basic aspects of Testing, but do not implement any automatic method.

PROCEDURE 6(a):

The system is tested outwards from the switching centre.

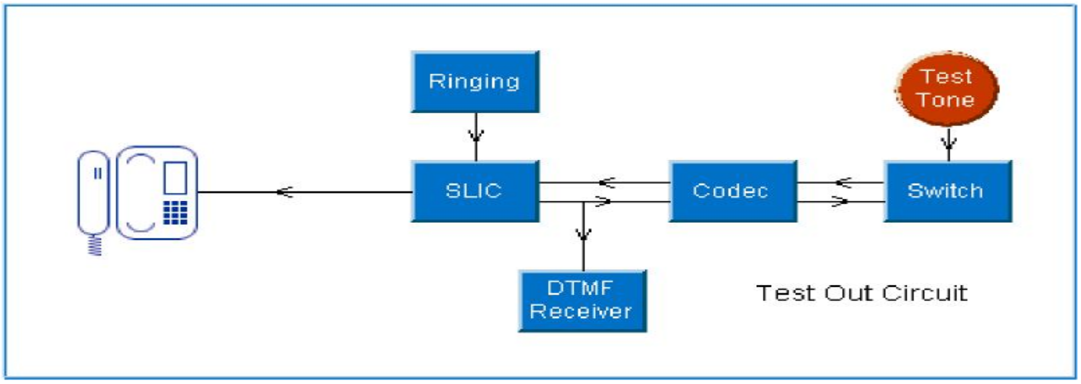

Figure 6.1 Testing circuit

A Test Signal is transmitted to a selected telephone. The Signal is one of:

1. Single Tone transmission test signal of 800 Hz;
2. Dual Tone signal, with 852 Hz and 1209 Hz signals combined;
3. The Ringing signal, used without a cadence.

The signals can first be detected by listening. It is not intended that there are any faults on the system, so all the signals should be heard.

Normal operation of the switch is suspended for this Assignment.

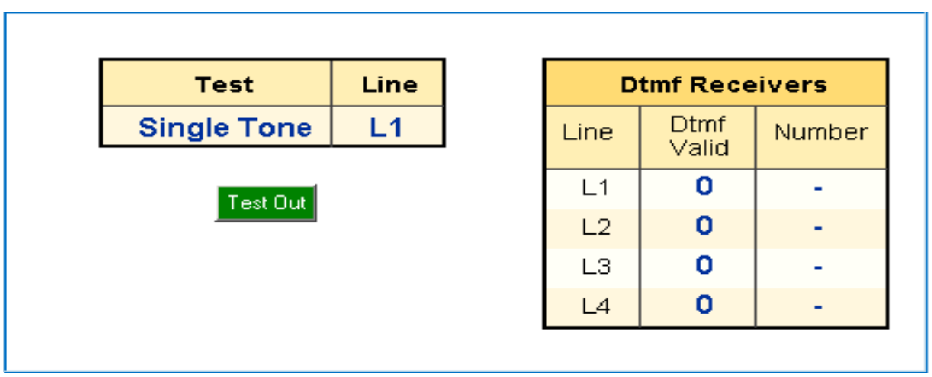

Figure 6.2 Status of DTMF Receivers

- Use the ***Test Out*** button to select the *Single Tone* and *Line L1,* and listen for it on Line 1. Is there a response by the DTMF receivers? Listen on different Lines.

- Apply the Ringing signal to different Lines, with the telephones On Hook.
- Select *Dual Tone,* listen for it, and observe the response at the DTMF receivers, with the telephones Off Hook.

QUESTIONS:

1. How can you be sure whether a DTMF Receiver has responded to the Dual Tone? Is it enough to see the number 7 as the received number?

2. If the Dual Tone is clearly heard at a telephone under test, but the DTMF Receiver for that telephone does not respond, where is there likely to be a fault?

3. If Ringing can be heard at a telephone, but the audio tones cannot be heard, and the DTMF Receiver does not respond, where could the fault be?

THEORY 6(b):

Loopback:

This is a facility of the Codec, which enables an input signal from the switch to be connected internally to the output and returned to the switch.

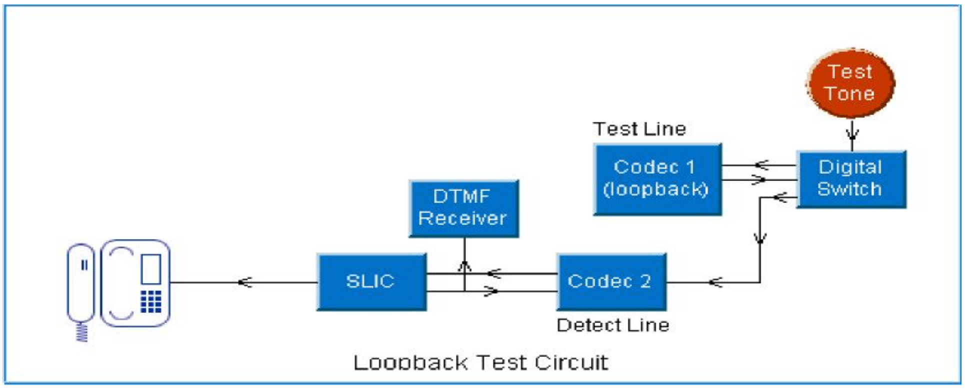

Figure 6.3 Loopback Testing

The Practical uses 2 stage switching to demonstrate the Loopback facility. The audio test signal is switched to the Test Line, from which it is returned by a Loopback setting at the Codec. The returned signal is then switched to a Detect Line, where it can be heard and also detected by the DTMF Receiver. The telephone for the Detect Line must be off hook.

Types of Loopback:

There are 2 forms of Loopback: Digital and Analog.

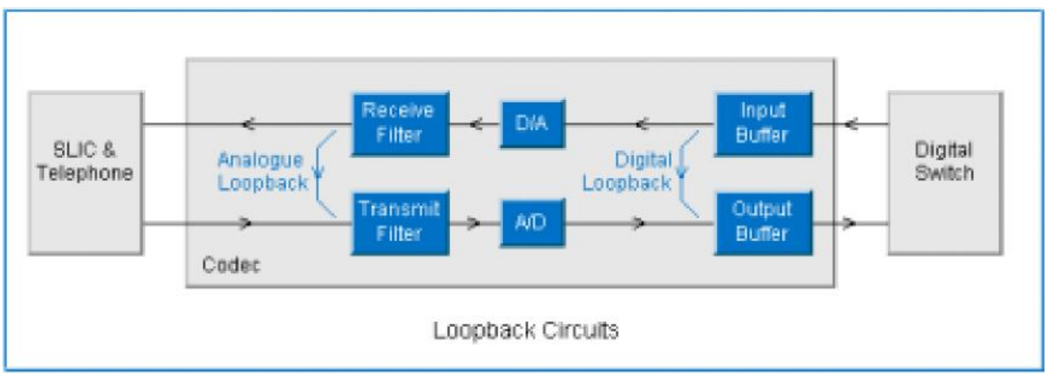

Figure 6.4 Digital and Analog Loopback Testing

The Digital Loopback takes the signal from the Input Buffer, in digital form and returns it to the Output Buffer. Thus the signal remains in digital form until its final conversion into analogue at the Detect Line. The Analog Loopback converts the signal into analogue form, and filters it. It is then returned to the Receive Filter, reconverted to digital form and returned to the Output Buffer. With either Loopback circuit, parts of the return path to the Switch can be tested. This is obviously useful for automatic system testing.

PROCEDURE 6(b):

In the Loopback test, a Tone signal is sent to a Codec in the Test Line. It is there sent back to the switch, which connects it to a Detect Line.

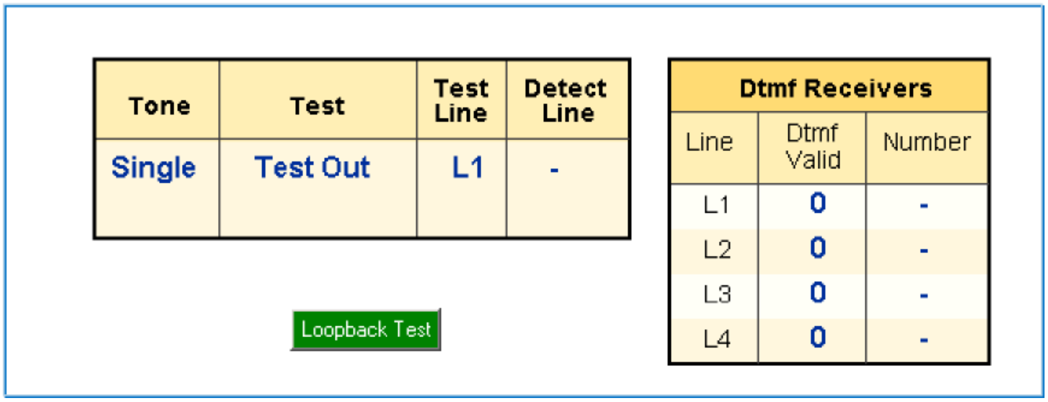

Figure 6.5 Status of DTMF Receiver for Loopback Testing

- First use the ***Single Tone*** and ***Test Out*** to *Test Line 1*, and listen at Line 1. Set ***Detect Line*** to any other Line. Then change to ***Digital Loopback,*** and determine where the tone can be heard.
- Use the ***Dual Tone,*** and observe the DTMF receiver response. Use other lines and the ***Analog Loopback*****.**

QUESTIONS:

1. What additional circuits can be tested by using the Loopback facility, compared with the Test Out?

2. If a Test Tone is heard clearly at a Test Line during Test Out, but not at the Detect Line during Loopback, where could the fault be?

3. What additional test would reduce the uncertainty of the above result?

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4. How could the analogue circuits of the Codec be checked?

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5. How could the testing arrangement of the system be improved?

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THEORY 6(c):

Telephone Traffic:

Telephone traffic is a measure of how much use is made of a particular telephone system, or even part of a system. The object of the analysis of traffic is to determine how much equipment is required to provide service for all users, without wasting resources on overprovision.

Traffic Calculations:

A basic characteristic of telephone traffic is its random nature. There are 2 quantities used to describe the nature of this traffic, both of them being random variables. Requests for calls arrive at random intervals. Over a long period this produces an average arrival rate R calls per second. Also each call has a holding time, which also varies randomly. The average holding time is h seconds.

The Amount of traffic carried in a period of time is the sum of the holding times for all the calls in the period. This is shown as Connected Time for each Line in this Assignment.

A more general expression for traffic is traffic intensity A or traffic flow. The Intensity is the *Connected Time* divided by the period of time in which it was determined. Traffic Intensity is therefore the average traffic.

Traffic has no dimensions, but usually has the unit of Erlang, named after a Danish mathematician. Alternatively it is expressed as hundred (century) call seconds per hour (CCS). 1 Erlang = 36 CCS, since there are 3600 seconds per hour.

The maximum capacity of one telephone channel is 1 Erlang assuming it is in constant use. Traffic Intensity is also equal to the product *A = Rxh Erlang*

Traffic Measurements:

The Assignment provides an introduction to the concepts of traffic analysis. However it is subject to some limitations:

1. With only 4 telephones, the maximum number of connections is 2, and therefore with the ample capacity of the digital switch there is no possibility of *blocking*, i.e. no call requests will be refused because of lack of capacity.
2. With any typical educational use, the telephones will be used to illustrate aspects of the control system, not to make normal calls. Hence the pattern of usage will not be typical. For example the amount of set up time may be much larger than for normal use.
3. The sample size is too small to be statistically significant.

System Traffic:

Telephone Traffic:

This is the use made of the telephone system. This Assignment shows how data is collected and calculated for establishing the traffic levels. The Practical uses the data accumulated by the Workboard since it was last started. The display shows the relevant data for each Line; and where appropriate the total values for the system.

Set Up Time:

The number of *Call Attempts* and the total time used for setting up calls are displayed. These are important data for the design of a telephone system, as it shows how much equipment is required Call Set Up.

Traffic Intensity:

This is the most widely used parameter for traffic, with the symbol *A*. It is the total usage of the system, the Connected Time during a particular period of time, divided by that period.

A = Connected Time / Total Time

Traffic Intensity has no dimensions, but is given the unit of Erlang.

The display shows the Connected Time for each Line. The Connected Time for the whole System is the sum of the Connected Times for each Line. From these is derived the Traffic Intensity for each Line and for the System.

Telephone Traffic:

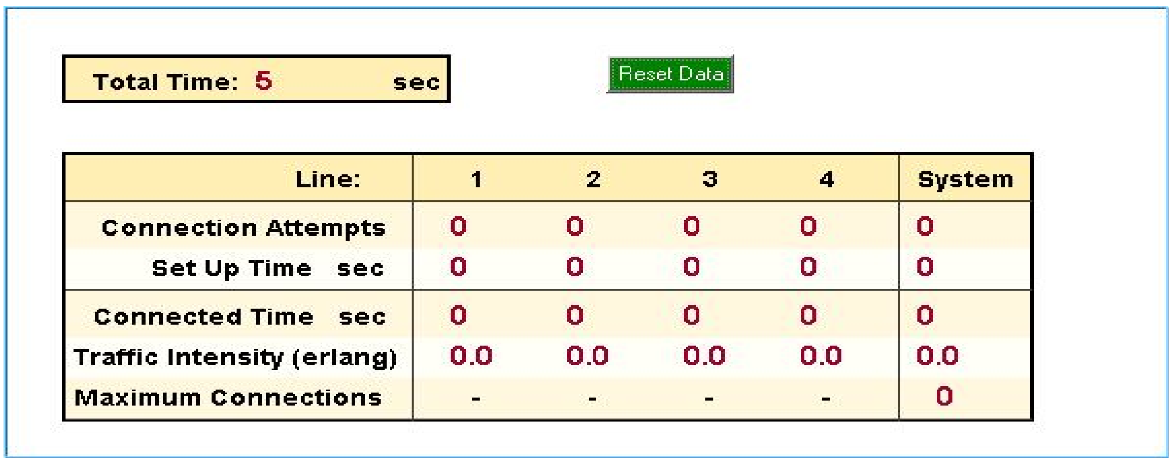
Finally, the *maximum number of connections* achieved at one time is shown. This cannot be large in this system, but is a very important parameter for design of large systems.

The data used for the traffic statistics in this system is collected each time a Practical is run, and is not lost until the system is switched off. However if it is desired, the data can be cleared to start again using the ***Reset Data*** button.

PROCEDURE 6(c):

The table shows the data used to calculate the traffic values for the system and the Traffic Intensity in Erlangs. The data has been collected over the period of use of the Workboard.

Make a connection between any two telephones.



Total Time: 5 sec

Reset Data

Line:	1	2	3	4	System
Connection Attempts	0	0	0	0	0
Set Up Time sec	0	0	0	0	0
Connected Time sec	0	0	0	0	0
Traffic Intensity (erlang)	0.0	0.0	0.0	0.0	0.0
Maximum Connections	-	-	-	-	0

Figure 6.6 Telephone Traffic Record

Observe the change in *Set Up Time* for the calling line when the call is answered. Observe the change in Connected Time and Traffic Intensity when the call is released.

QUESTIONS:

1. What is the maximum possible Traffic Intensity in Erlangs for this system?

2. If a Line has a Connected Time of 24 minutes in a measurement period of 1 hour, what is the Traffic Intensity for the Line?

3. If the 4 lines of the system have Traffic Intensities of 0.3, 0.25, 0.05 and 0.6 Erlang, what is the total Traffic Intensity for the system?

DUAL SWITCHING CENTRES

LAB SESSION# 07

TRUNK CONFIGURATION AND SWITCHING

OBJECTIVES:

- To evaluate the method of interconnecting two separate digital switches.
- To appreciate the directory numbering schemes for systems with many digital switches.
- To evaluate the sequences involved in making connections to the busses at the originating and destination switches.
- To analyze the use of the time slots employed in the multi-switch system.

Preliminary Procedure

Check that the Workstation is set up with two Digital Switch Centres and Telephone Trays (58-1 22 and 58-123). One Switching Centre Type should be set to 'A' and the other to 'B'. A 'curly' trunk cable should interconnect the 'Trunks' connectors. (If a Trunk Networks Board, 58-140, is included in the set-up this Assignment will automatically connect the trunks correctly via that board).

THEORY 7(a):

Digital Switch Allocation:

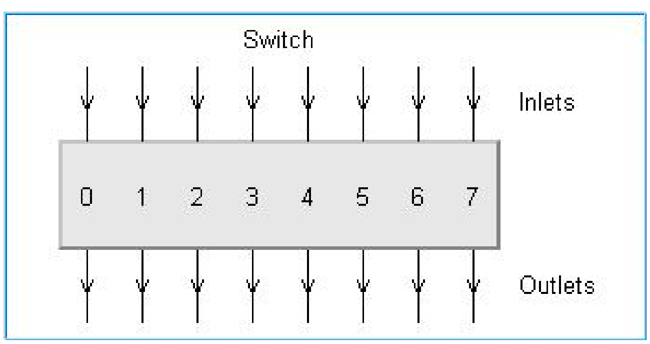


Figure 7.1. Digital Switch

The digital switch used for each Workboard has 8 Inlet Buses, and 8 Outlet Buses. Each Bus has 32 8-bit timeslots, and switching can occur between any of the timeslots.

For the Digital Switching Centre workboards the buses are allocated as follows:

- Bus 0 and 1: Connections to local telephones.
- Bus 2 and 3: Alternative trunk connections.
- Bus 4 and 5: Control of codecs to provide loopback testing in the Testing Assignment.
- Bus 6: Distribution of tones.
- Bus 7: Not used.

Dial Tone Production:

The dual frequency dial tone is produced by a combination of two devices.

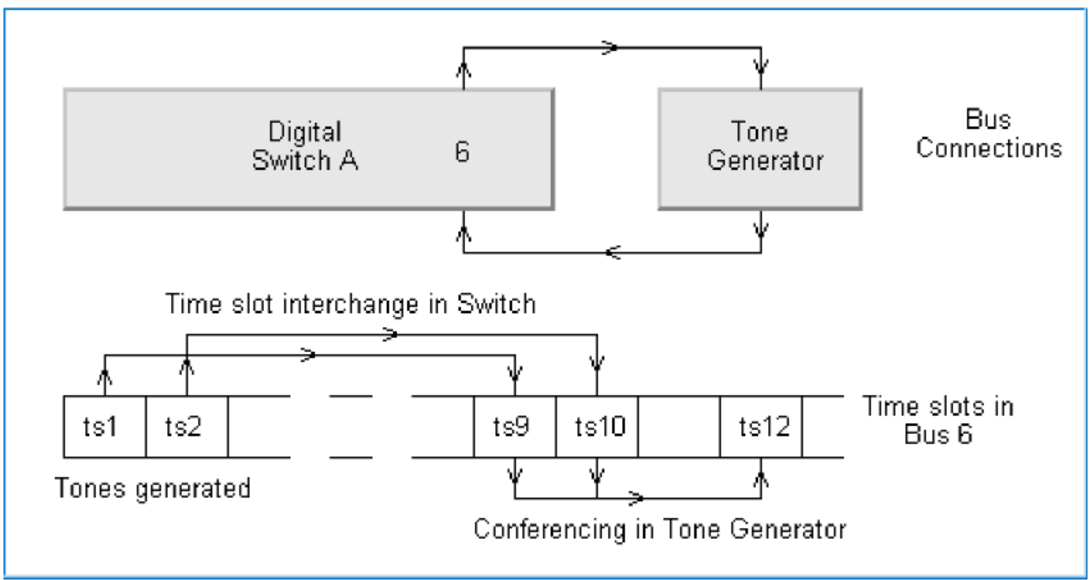


Figure 7.2 Time Slot interchange in Digital Switch

The principal device is a combined tone generator and conference circuit. This will produce individual tones in the range from 252 to 3969 Hz; at intervals varying from 3.91 to 31.25 Hz. Tones at these frequencies can be used directly for the simple tones such as Ring Tone or NU Tone. However, for the recommended Dial Tone, two tones at different frequencies need to be combined. The tone generator will also combine signals with the conferencing circuit. But it will not generate tones and conference them in the same time slot.

Therefore the two constituent tones are generated in timeslots 1 and 2, and transmitted to Inlet 6 of the Digital Switch on Workboard A. There they are switched to timeslots 9 and 12 in Outlet 6 which is connected back to the tone/conference circuit. The two tones are there combined with a zero input in timeslot 11, and output in timeslot 12. Finally that is connected to the Digital Switches, from which it can be switched to any telephone as required. The dual DTMF tone used in the Trunk Paths Practical for testing continuity is produced by a similar method.

Trunk Paths:

This is the first Assignment using 2 Switching Centres. The 2 Workboards are identical in construction, but are identified by different settings on the ***Switching Centre Type*** switch, at the far right hand corner. One board must be switched to **A** and the other to **B.** Calls can be made from either Switching Centre. For each call, the Switches are described as the *Originating Switching Centre,* and the *Destination Switching Centre,* respectively. An Inlet Bus and an Outlet Bus has to be designated in each Switching Centre for trunk connection to the other.

The trunk Outlet Bus from one Switch is connected to the trunk Inlet Bus at the other Switch by the curly **Trunk Cord.** The Cord has a *crossover* to provide the correct circuits between the Switches. The speech path for a trunk connection is provided by switching from a local Inlet to the trunk Outlet at one end and from the trunk Inlet to a local Outlet at the other. A similar procedure is used for the reverse direction.

The system can use either Bus 2 or Bus 3 for Inlet and Outlet at each Switching Centre. The user selects the Bus using the ***Bus Select Switches,*** and the selection is indicated by lamps on the Workboard. However there is no automatic method of determining the trunks selected. Therefore the Control needs to be instructed which position the switches have been set to, by clicking on the ***Set Trunk Paths*** button.

The system software is preset to Bus 2 for Inlet and Outlet on both Workboards. It will be found that speech contact can be made using different Busses for Inlet and Outlet on the same Switch. It would be normal practice to use the same Inlet and Outlet for a particular connection, but the architecture of the Switch allows other possibilities.

Continuity:

The Switch settings can be tested to ensure that the settings on the Workboards correspond with the values used by the Switch Control. It is the Speech path which needs confirming, and this can be provided by using a DTMF signal generated by the tone generator. Detection of the tone requires one of the telephones to be Off Hook, so that the DTMF receiver is able to receive the tone signal. Any telephone can be used, the system will find it.

The tone is connected to time slot 14 (ts14) on the Outlet trunk from the Switch with the telephone used for detection; and ts14 on the Inlet trunk is connected to that telephone. At the other Switch, ts14 on the Inlet and Outlet trunks are connected together, to form a loop.

The test signal can be heard if the connection is correct, and the Continuity:

- **Good** legend appears on the screen.
- Otherwise Continuity: **Fail** is shown.

PROCEDURE 7(a):

There are 2 possibilities for the Inlet and Outlet trunk connections on the Switching Centres A & B. The Inlet trunk can be connected to either Bus 2 or Bus 3 of the Switch by the ***Inlet Bus Select*** switch and the Outlet trunk has a similar choice. The continuity test checks the settings. The test uses a dual tone which can be recognized by a DTMF receiver. The test therefore requires that at least one telephone is off hook.

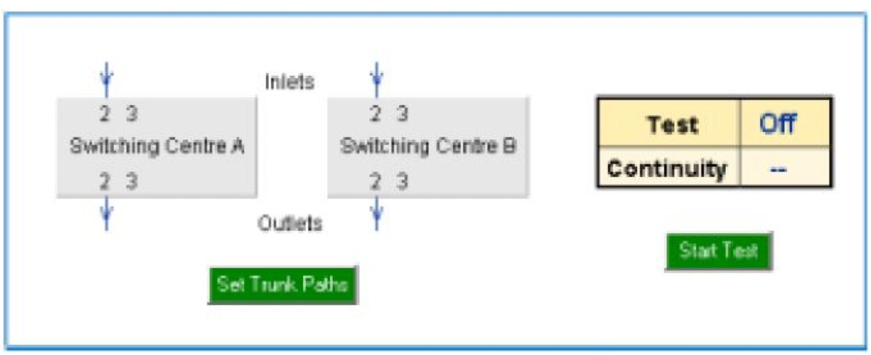


Figure 7.3 Inlet and Outlet Trunk Connection

Set the switches on both Switching Centres as required. Then enter those switch settings into the table, by clicking on the ***Set Trunk Paths*** button. Write down the settings of the switch:

Finally use the ***Start Test*** button to show that the Continuity and check if the settings are correct. Write your results:

These settings are retained until the power is switched off.

QUESTIONS:

1. With a 2 way connection, is it necessary for the Inlet and Outlet connections to each Switch to be made to the corresponding Buses?

2. If not why is it possible to use different Buses?

THEORY 7(b):

Directory Numbers:

The directory numbers used for each telephone and for each Switching Centre can be changed. Each Switch needs to distinguish between local calls and trunk calls. Hence the numbering scheme must allow for this discrimination.

- Each Switching Centre is allocated a single digit from 1 to 9.
- Each telephone is allocated a 2 digit number, which must not begin with either of the digits used for the

Switching Centres.

The preset Switching Centre identities are **4** ***(Switch A)*** and **5** ***(Switch B).*** The preset numbers for each telephone are the same as for the Single Workboard system, i.e. **21, 22, 31** and **32.** Any of the numbers can be changed, provided the rules are observed at all stages of the changes. The new numbers will be kept until the power is switched off. Now that the numbers have been set, the speech continuity established in the Trunk Paths Practical can be checked by calling from one Workboard to the other.

PROCEDURE 7(b):

- Preset Directory Numbers are provided for each Switching Centre, and for each telephone.
- These can be changed, using either ***Change Numbers*** button, but certain rules must be observed to match the protocol used by the Switch Control.

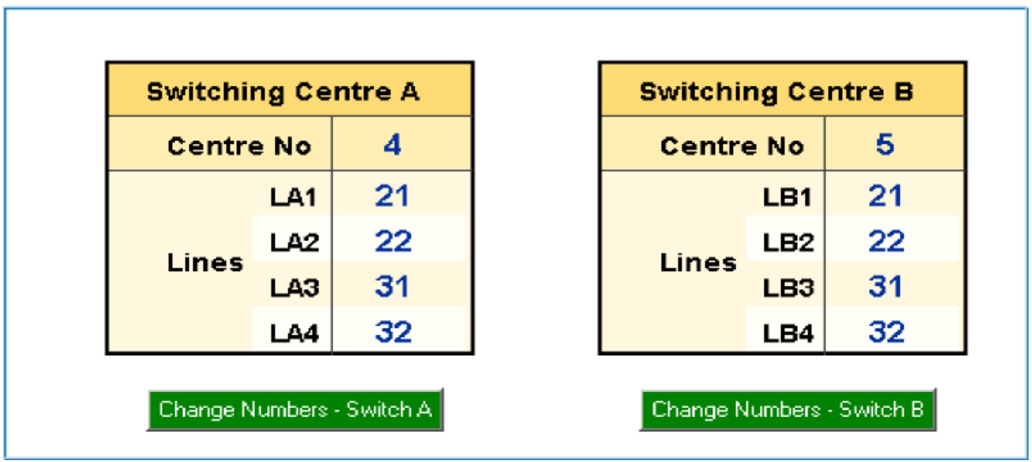


Figure 7.4 Directory Numbers of lines

1. The two Switching Centres must have different 1 digit identities.
2. The first digit of the 2 digit directory numbers for each telephone must not be the same as either Switching Centre identity.
3. The directory numbers within each Switching Centre must be different.

Whether the numbers have been changed or not, try out the system by making calls. Make calls within one Switching Centre, and then from one Switch to the other. Take observations.

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QUESTIONS:

1. Is it possible for the local directory numbers on both Switching Centres to be the same?

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2. Does the local Switching Centre number need to be used for a local call? Can it be used?

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3. If the same directory number is used on each Switching Centre, how is confusion avoided?

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4. Why are some numbers not allowed for the local directory numbers?

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THEORY 7(c):

Trunk Switching:

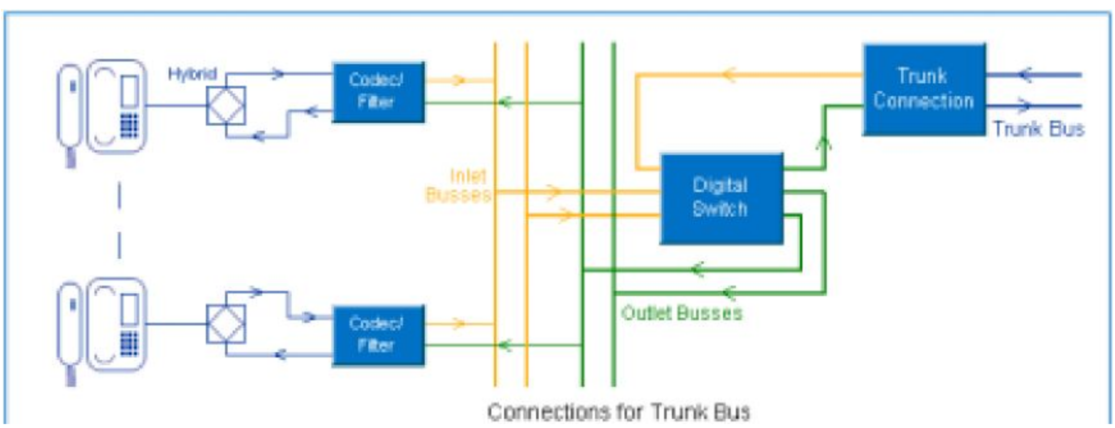

Figure 7.5 Trunk Switching

The local Inlet and Outlet Busses are connected by each Digital Switch to the Trunk Bus. The Trunk Bus connects the two Switches so that the Outlet Trunk Bus from one Switch is connected to the Inlet Trunk Bus at the other. Thus each Digital Switching Centre can switch both local and trunk calls. In the experimental equipment the Trunk Cord connecting the 2 workboards carries the Trunk Bus.

Long Distance Transmission:

The waveforms transmitted between the two Workboards only have a short distance to travel, and thus there is no deterioration in the waveform. The waveform used for the transmission is thus the basic Non-return to Zero (NRZ) Unipolar signal used by normal logic circuits. However, in commercial telephone systems, the digital waveforms must be transmitted over long distances at a high bit rate.

Typical limitations of long distance cables are:

1. An inability to transmit dc levels. They are often transformer coupled, and the lines may be used to send dc power to repeaters.
2. Limited bandwidth due to the capacitance of the cable.
3. The lack of a separate channel for carrying timing information, which is needed to keep the transmitter and receiver in synchronism.

Therefore signal waveforms are designed to overcome these limitations. The methods used are known as line coding.

Substitution Codes:

In addition to Line Coding, various substitution codes are used to avoid long strings of zeroes. If a string of zeroes does occur, a special code is used to replace it. The code is recognized because the regular AMI sequence is violated. A typical code is the *B3ZS* (binary 3 zero substitution) code, used in T1 systems. In this any sequence of 3 zeroes is replaced by either B0V or 00V. B is a correct '1' pulse and a V is a violation '1' pulse. The sequences are selected so as to avoid a net dc value. A code used in CEPT systems is *HDB3* (high density bipolar coding). Any sequence of 4 zeroes is replaced by sequences with a violation in the last bit position.

Outward Path:

The connection between the two Switching Centres is made by a dedicated 30 channel bus, the Trunk Bus. This requires circuits for both directions, outward and Return. This Practical looks at the Outward Path. Switching occurs in two stages in each switch:

1. The internal bus in the Originating Switching Centre, is connected to the trunk bus.

2. The other end of the trunk bus is connected to the internal bus at the Destination Switching Centre.

For the Practical, Switching Centre A is used as the Originating Switch, and Switching Centre B is the Destination Switch. As soon as dialing is complete, if the required line is available, the internal bus in Switch A is immediately connected to the trunk bus. The connection in Switch B is only made when the telephone is answered.

Four timeslots in the trunk bus are allocated for transmission in each direction. They are recorded in a busy time slot map, similar to the busy line map in the Line Maps Assignment in the Single Switching Centre package.

PROCEDURE 7(c):

Demonstration of the switching required for the Outward path for calls from Switching Centre A to Switching Centre B. The Trunk Bus is connected from an Outlet of Switch A to an Inlet of Switch B using the Outlets and Inlets previously selected.

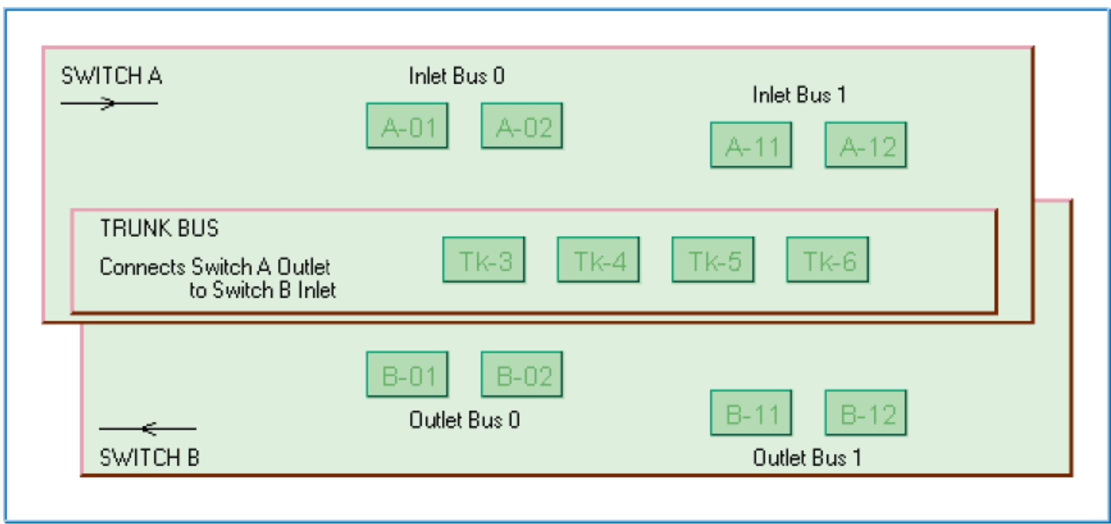

Figure 7.6 Outward Path Switching

Make a connection from a telephone on Switching Centre A to one on Switching Centre B. Observe the time slot allocated in the **Trunk Bus** for the connection, and when the connections are made. Repeat and have several connections at one time.

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Connections in the reverse direction, from Switching Centre B to Switching Centre A use timeslots Tk-7 to Tk-10.

QUESTIONS:

1. When is the connection made to the internal bus in the Destination Switch?

2. Why is that connection not made immediately dialing has finished?

3. Why are 4 timeslots allocated to each Switching Centre for the Trunk Bus?

4. Why are separate groups of timeslots used for calls starting at each Switching Centre?

5. In what order are the trunk timeslots allocated to the calls?

THEORY 7(d):

Return Path:

This Practical shows the Return Path for connections in the same direction as in Practical 1. Switching Centre A is still the Originating Switch, and Switching Centre B is the Destination Switch. When dialing is complete, if the line is available, the connection is immediately made in Switch A between the trunk bus and the internal bus. In Switch B, the Ring Tone is connected to the trunk bus, using the same cadence as the Ringing applied in Switch B to the required line. When the telephone is answered, the internal bus is connected to the trunk bus in place of the Ring Tone.

PROCEDURE 7(d):

Demonstration of the Return Path switching for calls from Switching Centre A to Switching Centre B.

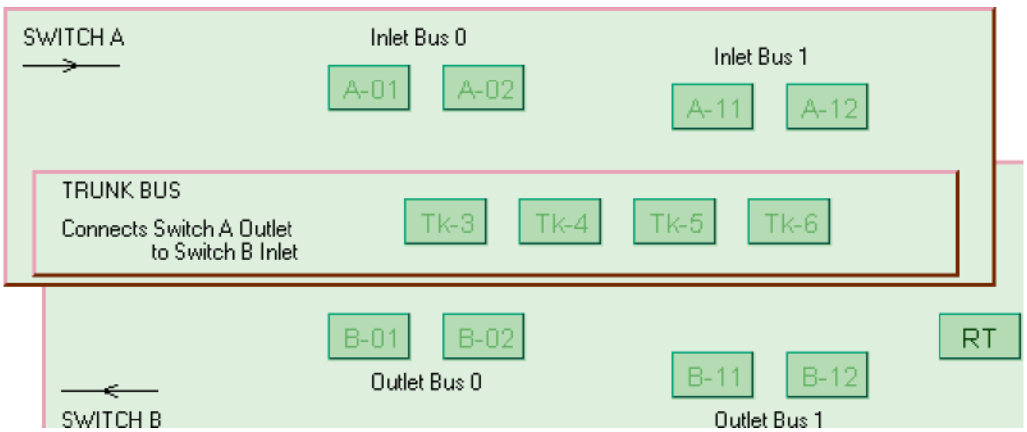

Figure 7.7 Return Path Switching

Make a connection from a telephone on Switching Centre A to one on Switching Centre B. Observe when the Ring Tone is connected and disconnected. Repeat and have several connections at one time.

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Connections in the reverse direction, from Switching Centre B to Switching Centre A use timeslots Tk-7 to Tk-10.

QUESTIONS:

1. Explain the sequence of switching, at both switches, to enable Ring Tone to be returned from the Destination Switching Centre.

2. Does the system use the same timeslot number for transmission in each direction for one call. Is this physically necessary?

3. How many active connections are needed for a trunk connection between Switching Centres A and B, to cater for connections originating on either Switch and allowing speech paths in both directions?

LAB SESSION# 08

TRUNK SIGNAL UNITS

OBJECTIVES:

- To analyze the trunk signaling in terms of the International Standards Organization (ISO), Open Systems Interconnection (OSI) model.
- To investigate the structure and function of Message Signaling Units (MSU) and Fill In Signaling Units (FISU) and how to differentiate between them.

THEORY 8(a):

ITU-T Signaling System No 7:

The ITU-T Signaling System No 7 (SS7) is the current ITU-T standard for international telephony. It is widely used for Common Channel Signaling. However it is too extensive and complex to be used directly in these Assignments. Therefore a simple signaling system has been devised, using the structure and terminology of SS7 where relevant. The structure of SS7 is a series of Levels, from 1 to 4. Each Level accepts data from a higher Level at the transmitting end, and has no interaction with that data. At the receiving end, each Level responds to the action of the equivalent Level at the transmitting end, and sends the remaining data to the next higher Level.

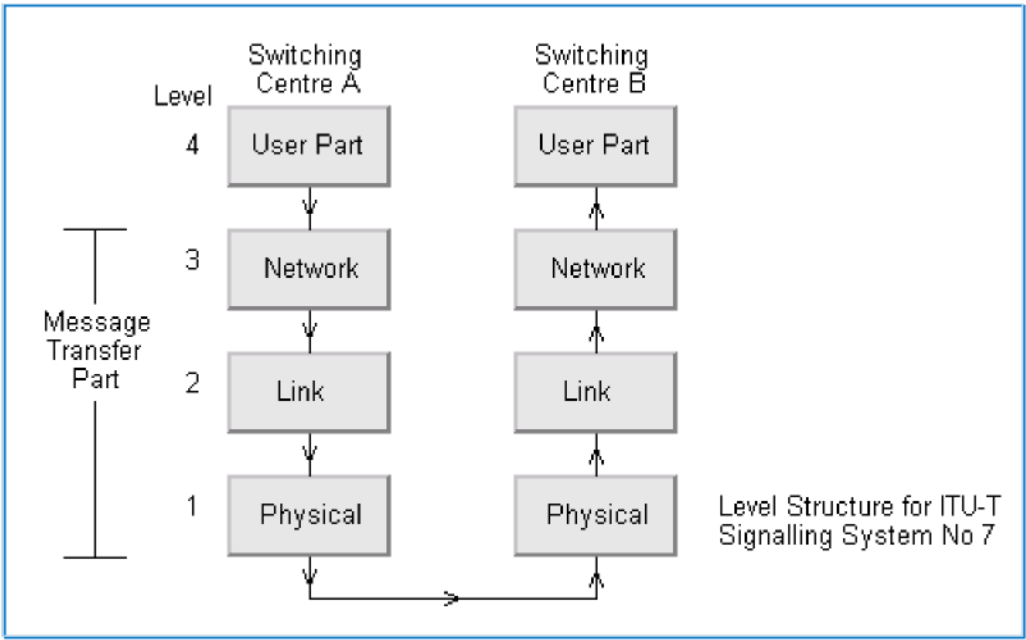

Figure 8.1 Level Structure of SS7

A message is shown passing from Switching Centre A to Switching Centre B. The first 3 Levels, the Message Transfer Part, deal with efficient and reliable transmission of Messages.

Levels:

The levels of SS7 can be described as follows:

- **Level 1 – Physical -** deals with the physical transmission of the data. In this system the use of timeslot 16 is a Level 1 specification. It defines the transmission rate at 2.048 Mbps, and the effective signaling rate as 64 kbps. Other physical parameters such as voltage levels and line coding used are also matters for Level 1.
- **Level 2 - Data -** controls the Data Link. The use of Flags as delimiters, the Sequence Numbers, Length Indicator, Message Signal Units, Fill In Signal Units and the Frame Check Sequence FCS are all defined in Level 2. The use of the Sequence Numbers and FCS for Error Control are typical activities of the Level structure. They are added to the data in Level 2 at the transmitting end, and checked and removed in Level 2 at the receiving end.

Level 1 just accepts them, and transmits them. Level 3 is unaware of them, but assumes that Level 2 will provide a reliable transmission service. It does not need to know how this is performed.

- **Level 3 - Network -** controls the routing of the control signals. In a large network there are different possible routes for Messages to get to the correct destination, and these are managed at Level 3. In this dual Switching Centre system there is only one route, there and back, so there is negligible action at this Level.

- **Level 4 - User Part:** Level 4 in this case is a Telephone User Part (TUP). Alternatives are the ISDN user part (ISUP), a Data User Part (DUP) and a Transactions Capability Part (TCAP). The Messages to be sent in the Signaling Information Field are defined at Level 4. They will be explored in the Signaling Information Field Assignment.

Open Systems Interconnection ISO-OSI model:

Since SS7 was defined, the International Standards Organization (ISO) has used the same principle to define the OSI for communication between computers. It uses 7 'layers' instead of 4 Levels, to allow for the complexity of the data structures in computers. With some additions to Level 3, the bottom 3 layers of the OSI model correspond to the MTP (Levels 1 to 3) in SS7.

In modern practice, SS7 is used for circuit related activity, and OSI for other requirements.

Message Signal Units:

Message Signal Units (MSUs) are the basic method of sending signals from one Switching Centre to another. Like all Signal Units, they start with 2 Sequence Numbers, the Forward Sequence Number FSN and the Backward Sequence Number BSN.

The FSN increases by one each time a new message is sent. If the message is received correctly, then the BSN in the reverse direction is increased to be the same. This indicates to the sending Switching Centre that the message has arrived correctly. Thus the backward numbers BSN are concerned with transmission in the reverse direction!

The Sequence Numbers cannot increase indefinitely. The maximum value is 15 in these Assignments, after which the next Number is zero. Following the Sequence Numbers is the Length Indicator *LI*. The LI simply states how many bytes of data follow in the Signaling Information Field. Obviously if there is a message, there must be more than 1 byte. But also the formal definition of an MSU is that the LI is greater than 0. The reasons for the different values of LI are covered in the Assignment on the Signaling Information Field.

PROCEDURE 8(a):

Each Signal Unit (SU) carries 2 Sequence Numbers. The Forward Sequence Number (FSN) is increased by

1 for each new message. If the message is received correctly, the Backward Sequence Number (BSN) for SUs in the reverse direction equals the received FSN. Hence the BSNs refer to transmission in the reverse direction.

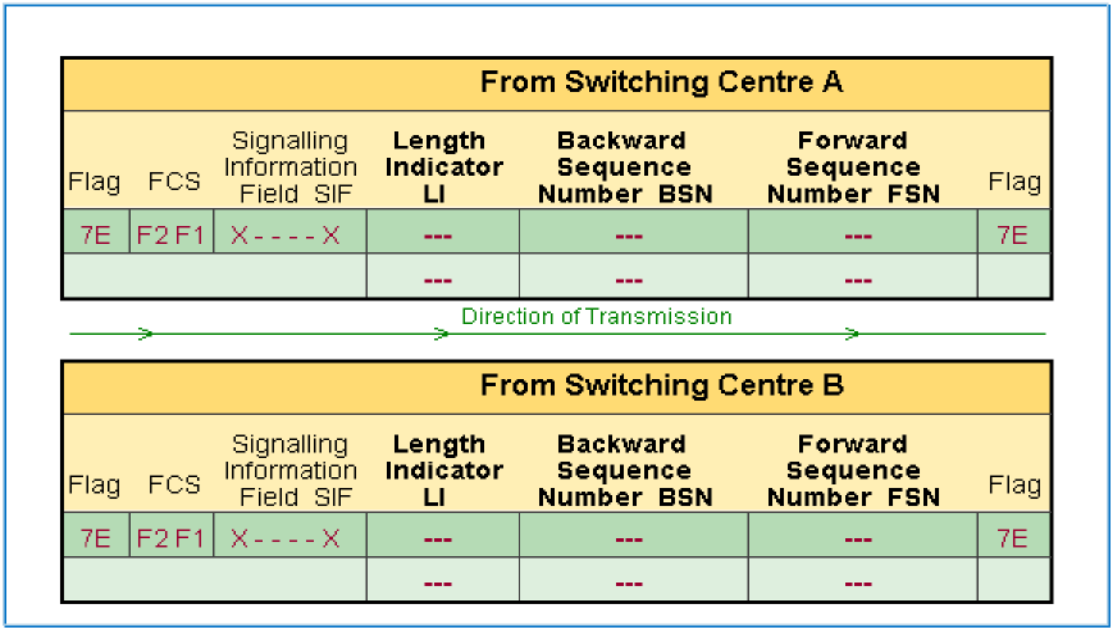

Figure 8.2 Signaling Unit Structure

The Length Indicator LI shows how many bytes of data follow. If LI is greater than zero, then the Unit is an MSU.

Make trunk calls in both directions and observe the LI values for different messages, and the operation of the Sequence Numbers.

QUESTIONS:

1. How is a Message Signal Unit distinguished from another Signal Unit?

2. What values of Length Indicator LI occur for:

(i) Starting a trunk call;

(ii) Clearing a trunk call;

(iii) Responding to an attempted trunk call to an unallocated number.

3. Where is the information carried in a Message Signal Unit?

THEORY 8(b):

Fill In Signal Units:

Fill In Signal Units (FISUs) are used if there are no messages to be transmitted. They contain Forward and Backward Sequence Numbers, identical to those in the Message Signal Units. In fact that is an important reason for the use of FISUs. A transmitting Switching Centre does not know if its MSU has been correctly received until the BSN in the reverse direction is seen. It cannot reasonably wait until there may be a message in the reverse direction. Therefore a Signal Unit is sent in each direction every 20 ms. If there is a message waiting, then an MSU is sent. But if not, then an FISU is sent.

Since there is no message, there is no Signaling Information Field, and the Length Indicator LI is zero. An

FISU is defined by having the LI = 0. The FISUs also check that the trunk connection is physically intact. If the trunk 'Cord' between the Workboards is not in position, then the FISUs are not received. The status of the Cord is monitored during the Practicals, and an error message appears if its absence is detected by the lack of Signal Units.

The Protocol Controller PC is connected directly to the trunk Cord. During timeslot 16, the PC is enabled, and the Switch Outlet is disabled. However, the operation of the Message system is no check that the speech path is secure. If the Bus Select switches do not correspond with the settings encoded in the Trunk Paths Assignment, then the speech path is not complete.

PROCEDURE 8(b):

Messages are sent from one Switching Centre to the other at irregular intervals. However, the message system requires that the Sequence Numbers are returned promptly so that the sending Switch knows whether a Message SU has been correctly received. Therefore Signal Units are sent regularly, every 20 ms, from each Switch. If there is no message waiting, a Fill In Signal Unit FISU is sent.

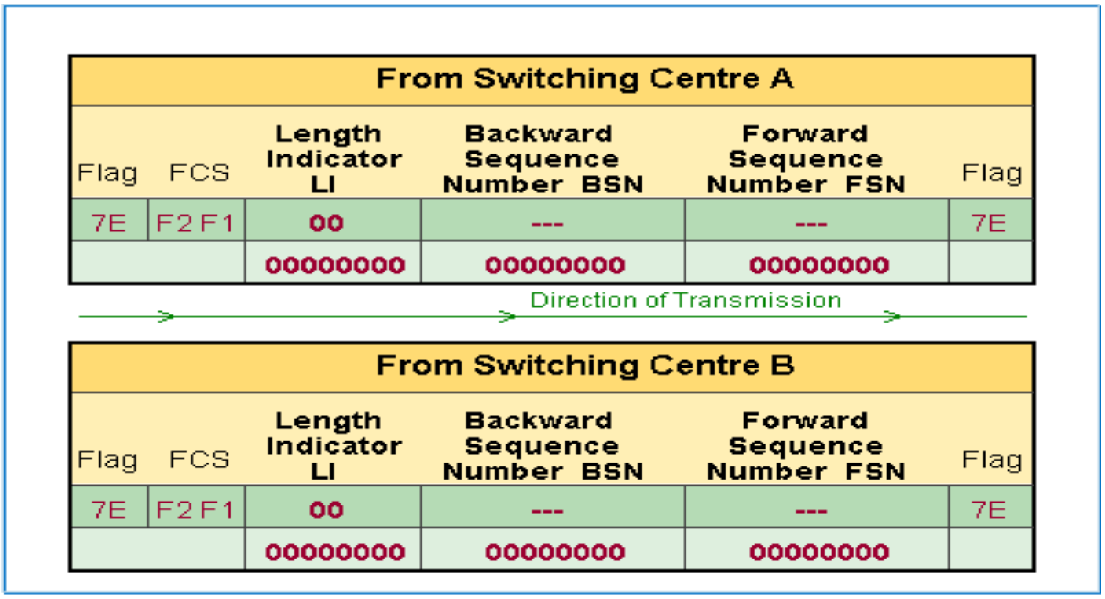

Figure 8.3 Fill In Signal Unit

An FISU is recognized because its Length Indicator is equal to zero. The regular sending of SUs also checks whether the connection is physically intact.

Make trunk connections and observe the Forward and Backward Sequence Numbers. The use of Sequence

Numbers is further considered in the Error Control Assignment.

QUESTIONS:

1. What is the maximum value of Sequence Number used?

2. Is zero a valid Sequence Number?

3. What is the value of the Length Indicator in an FISU?

LAB SESSION# 09

SIGNALING INFORMATION FIELD

OBJECTIVES:

- To investigate the format of a trunk signaling system based on the principles of the Signaling System No. 7.
- To determine the methods used to identify originating and destination switch centres, the time slot used and the purpose of the message.

THEORY 9(a):

Signaling information Field:

The format used for the Signaling Information Field of Label, then Heading Code then Address Signals corresponds in principle to the format of Signaling System No 7 (SS7). However, the individual components are reduced in size and scope.

In SS7 the Destination Point Code and the Originating Point Code each take 14 bits. Then the Circuit Identification Code CIC requires 12 bits. Thus the Label requires a total of 5 bytes instead of 2 used here.

The SS7 requires the use of the least significant 5 bits in the CIC to define the timeslot used, just as in this system.

The Heading Codes used are identical to the SS7 codes. There are however many codes not used. For example with long international dialing codes, it can be useful to start transmitting the number before dialing is complete. In that case the Initial Address Message IAM is followed by one or more Subsequent Address Messages (SAM).

When enough digits have been received, the Destination Switching Centre can respond with Address Complete (ACM code = 31 in hex). ACM is in the Successful Backward Message group, with the hex code 14.

Also an IAM or a SAM has to specify how many Address Signals are being transmitted. Other codes cover topics such as charging, the nature of circuit requested (e.g. no satellite, or all digital), continuity checks, congestion and maintenance.

Label:

The Label is the first item in the Signaling Information Field.

There is a unique Label for each attempt at a connection, whether it is successful or not. It consists of 2 bytes:

1. The Point Codes PC contains two codes for the Originating and Destination Switching Centres (SCs). First is the Destination Point Code DPC. This consists of 4 bits defining either of Switching Centres A or B. The hex codes for A and B are used.

Then the Originating Point Code OPC is another 4 bits, similarly defining the Destination.

Since there are only the 2 Switching Centres, the only possible PCs are AB and BA. Remembering that the least significant bit is transmitted first, in hex code the Label for a call from A to B is AB.

2. The Circuit Identification Code CIC defines the circuit at the Originating Switching Centre which is used for the connection. The first (least significant) 5 bits define the timeslot which is being used. They are allocated by the Originating SC. The final (most significant) 3 bits define the Outlet bus used by the Originating SC. This the Outlet defined in the Configuration Assignment.

Each Message about the connection, whatever the direction of the Message, uses the same Label. It identifies the connection the Message is concerned with.

PROCEDURE 9(a):

The Label is the first part of each Message Signal Unit (MSU). It defines the connection which is the subject of the MSU. Every MSU concerned with that connection has the same unique Label.

The Label consists of 2 parts; the Point Code byte PC and the Circuit Identification Code byte CIC.

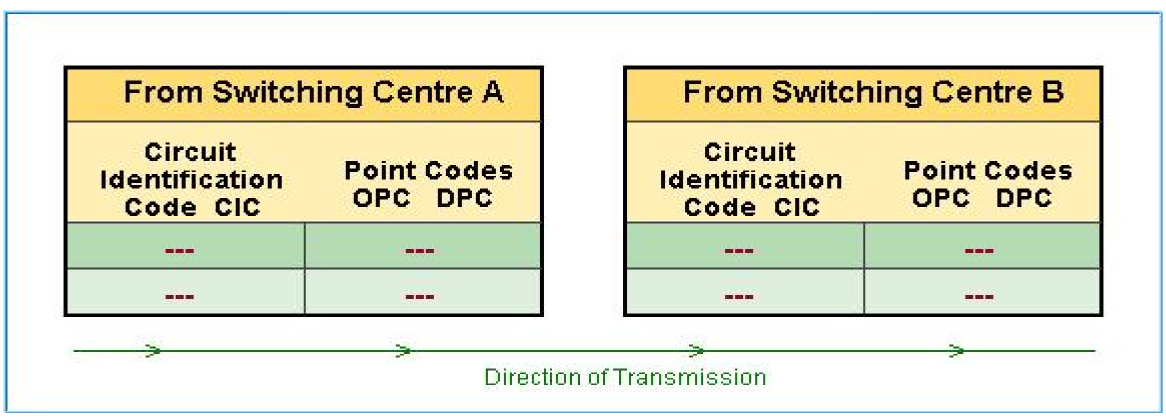

Figure 9.1 Label Elements

The Point Code starts with the Destination Point Code DPC, either the ASCII codes for A or B, and then follows the Originating Point Code OPC. The Circuit Identification Code has 5 bits to identify the timeslot being used for the connection, and 3 bits for the bus. The bus is 2 or 3 as defined in the Configuration Assignment.

Make trunk calls and observe the elements of the Labels at each stage and for each direction.

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QUESTIONS:

1. If a Label starts with the Point Code AB, what information does it convey?

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2. Are MSUs with the Point Code AB always transmitted in the same direction?

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3. If a Circuit Identification Code CIC is 84, what bus and what timeslot is being used?

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THEORY 9(b):

Heading Codes:

The Heading Code states the purpose of the Message Signal Unit. Each code has a 4 bit H0 field, followed by a 4 bit H1 field.

- The H0 field defines which group a Message belongs to. For example 0110 group is Call Supervision Messages.
- The H1 field defines the purpose more exactly. For example 0111 is for Calling Party Clear, known as CCL.

Thus the Heading Code for CCL is 76 in hex. Only 5 Heading Codes are required for this system, out of about 50 defined in SS7.

PROCEDURE 9(b):

The Heading Codes are transmitted immediately after the Label. They state the purpose of the current MSU.Only 5 Heading Codes is used in this system, out of a total of about 50 in the ITU-T Signaling System No 7 (SS7).

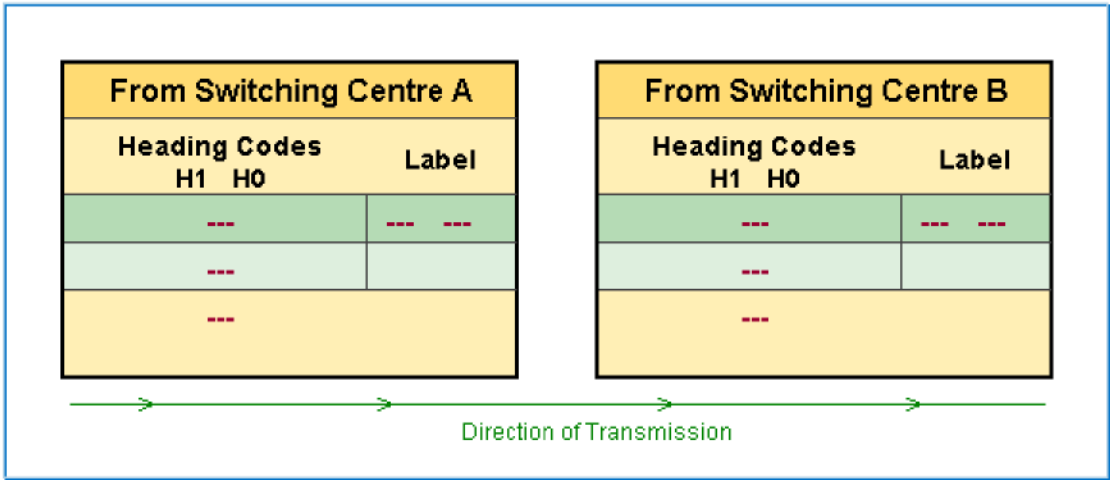

Figure 9.2 Heading Codes Information

They consist of two 4 bit codes. The first is H0, which defines which group of messages the current MSU belongs to. The second Heading Code is H1 which defines the particular code.

Make successful and unsuccessful trunk connections and identify all 5 Codes.

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QUESTIONS:

1.What is the sequence of Heading Codes used to establish and clear a successful connection?

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2.What Codes are used for unsuccessful attempts at connection?

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THEORY 9(c):

Address Signals

The Address Signals are only required for the Initial Address Message, which is the first Message when setting up a connection.

The first dialed digit specifies the Destination Switching Centre. That information is encoded in the Destination Point Code of the Label and therefore does not need to be included in the Address Signals.

The next 2 dialed digits form the Address Signals. Only the Destination Switching Centre has the information about the Numbering of its Lines. Therefore only at the Destination can the Required Line Identity be determined from the dialed digits.

The Signaling Information Field of a Message Signal Unit thus consists of a Label, Heading Codes and optional Address Signals.

PROCEDURE 9(c):

When a trunk connection is first set up, the Initial Address Message IAM contains the dialed numbers of the required line. Since the local numbers have two digits, the Address Signals contain the two dialed numbers DA1 and DA2.

This completes the Signaling Information Field SIF, which comprises the Label, the Heading Codes and the Address Signals. The Length Indicator is also shown for completeness.

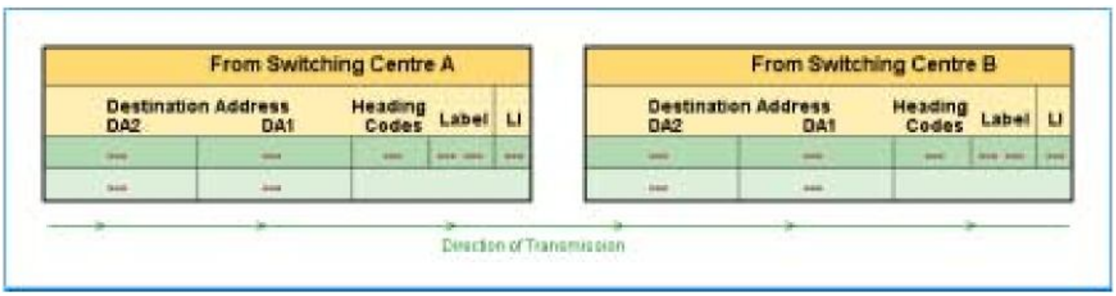

Figure 9.3 Addressing Signals

Make trunk connections and observe which digits are transmitted, and the complete SIF.

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QUESTIONS:

1.A trunk connection requires 3 digits from the user. Why are only 2 digits transmitted to the Destination Switching Centre?

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2.Why are the dialed numbers transmitted, and not the Required Line Identity?

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3.If Address Signals are not included in MSUs after the Initial Address Message (IAM), how do the Switching Centres relate the successive MSUs to the correct connection?

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LAB SESSION# 10

STATE DIAGRAMS

OBJECTIVES:

- To analyze the trunk call set-up process using the Call State Transition Diagram (CSTD).
- To evaluate the sequence of states and signals associated with a multi-centre switching system
- To familiarize with the relationship between information held on records and the description presented by the CSTD.
- Appreciation of the relationship between CSTDs for several switching centres concerned with the connection of a particular call.

THEORY 10(a):

Call State Transition Diagrams:

CSTDs for telephone control were introduced in the single Switching Centre Assignments. They are a subset of the ITU-T Specification and Description Language SDL. Each CSTD describes the operation of a specific computer process. In this Assignment the CSTD concept is developed by considering interactions between different processes, described by different CSTD.

The interaction between a main process and a subordinate one is shown. The *Originating Switch* CSTD is at the main level of operation. When it is in State S2, control is transferred to the *Dialing* CSTD. The *Dialing* CSTD is subordinate to the main CSTD. It keeps control until dialing is complete, when the result is returned to the *Originating Switch* CSTD. The combination of CSTDs enables complex systems to be specified in simple terms, with complete reliability.

Also two CSTDs at the same level are seen interacting with each other. The CSTDs for the Originating and Destination Switches cannot operate in isolation. They exchange Messages with each other outside the immediate Switching Centre where each operates. They use the SS7 Message system for communication.
Each Switching Centre uses both Originating and Destination processes, as calls may be initiated at any Switch.

SS7 Level 4:

The processes defined by the CSTDs operate at Level 4 of the ITU-T Signaling System No 7 (SS7). Each CSTD defines a Message to be transmitted to the other Switching Centre, e.g. the Initial Address Message IAM. It does not define in any way how the Message is to be transmitted. It just assumes that there is a system which will provide reliable transmission. The system is provided by the lower levels. It is also interesting to note that SS7 itself is defined using SDL process diagrams, similar to the CSTDs used here.

CSTD for Originating Switch:

The Originating Switch for Trunk calls must handle both local and trunk calls. The computer process which controls the combined function is defined by a new Call State Transition Diagram.

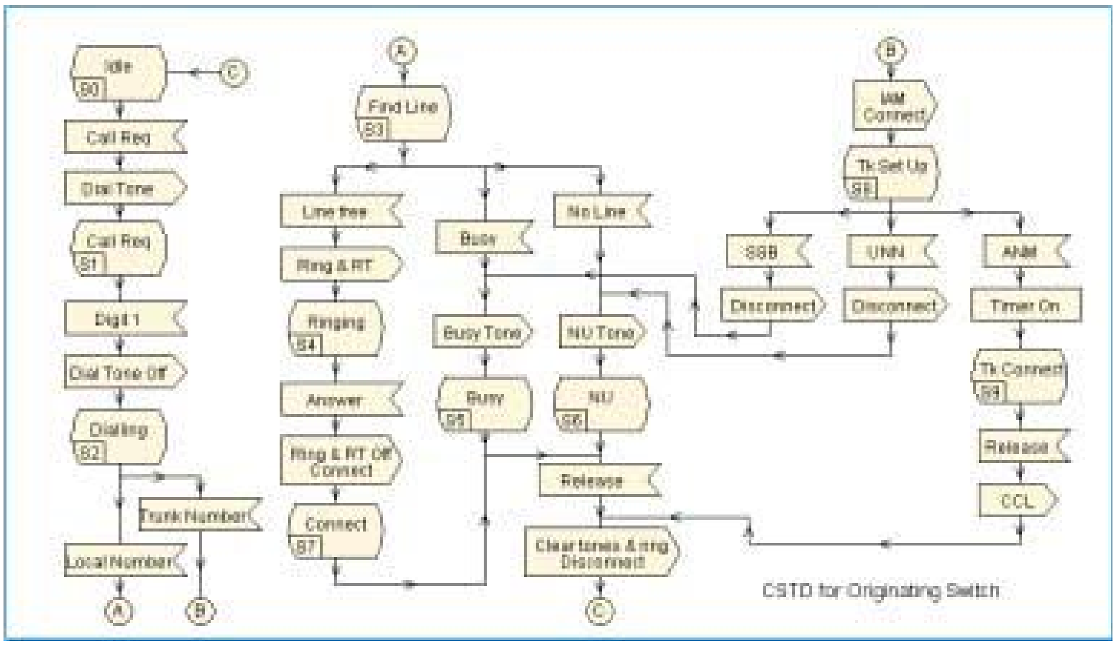

Figure 10.1 Call State Diagram

Local calls are treated in a similar manner to those in the Assignments which only had local calls. As before, state 3 (Find Line) is an internal transition which is passed through rapidly. However, for trunk calls, the CSTD provides 2 additional States, S8 (Trunk Set Up) and S9 (Trunk Connect).

Also the directory numbers may be longer, and they are now controlled by a subsidiary process *Dialing*. That process is defined by an independent CSTD, examined in a separate Practical. While the Dialing process is in operation, the Originating Switch process remains at State S2. When the dialing is complete, the next State depends on whether the Dialing process has detected a local or trunk number. If a trunk connection is requested, the process waits in State S8 until a response is received from the Destination

Switch. If the response is positive (Message ANM) then the timer is started, and a transition is made to State S9. Otherwise the process reverts to the appropriate State - Busy or NU.

PROCEDURE 10(a):

The Originating Switch must handle both local and trunk calls. Hence the Call State Transition Diagram (CSTD) has two more States S8 and S9 in addition to those for purely local calls.

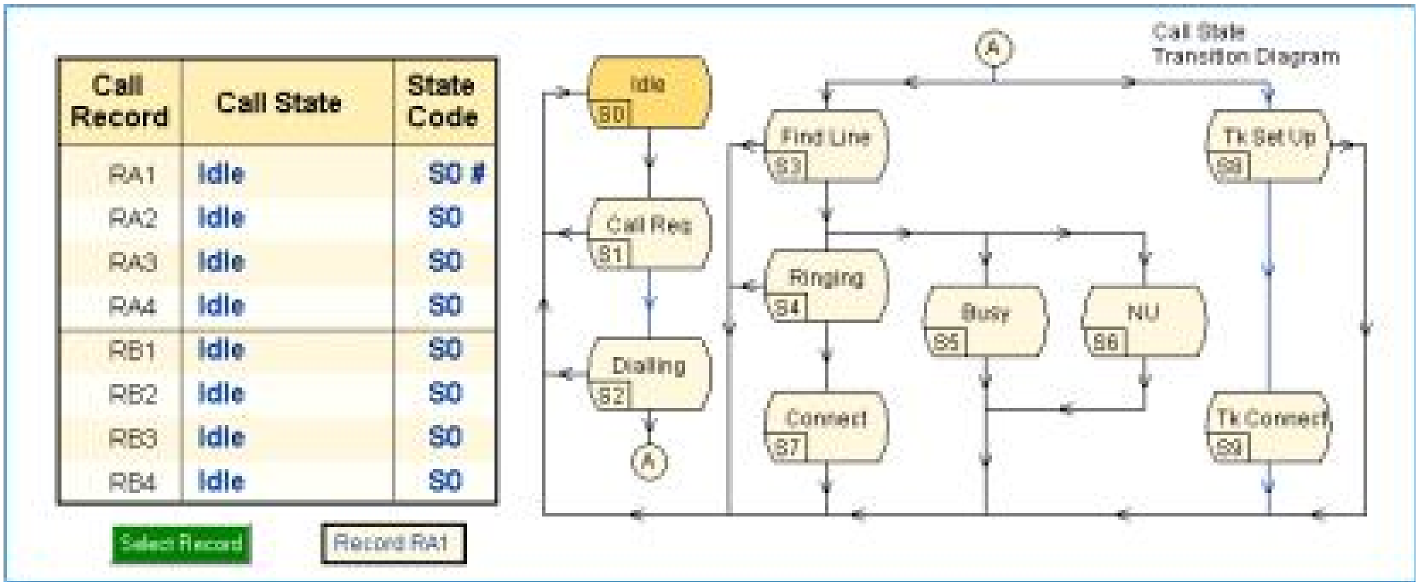

Figure 10.2 CSTD and Call Record

The table shows which State each Call Record has reached. The related Call State Transition Diagram (CSTD) shows the progress of the call. If necessary use the ***Select Record*** button to find the current CSTD. (# shows which one is displayed.) Make local and trunk calls and observe the States used. Record your Observations:

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Make trunk calls to Busy & NU numbers.

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QUESTIONS:

1. What additional States are used for trunk calls?

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2. Why is there no separate State for Trunk Ringing in this CSTD?

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3. At what stage of a Call is the Transition made from State S8 (Trunk Set Up) to State S9 (Trunk Connect)?

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THEORY 10(b):

Dialing CSTD:

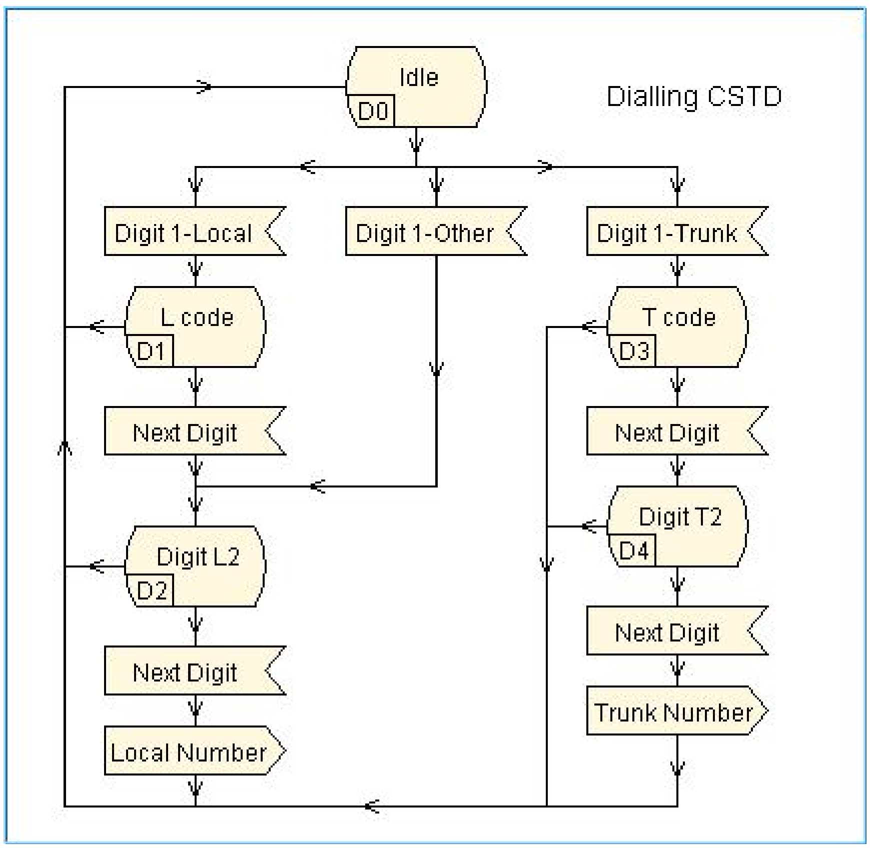

Figure 10.3 Dialing CSTD

The Dialing process is activated when the Originating Switch CSTD reaches state S2. It compares the first number received with the stored Directory Numbers for the 2 Switching Centres. If they correspond to either, then States D1 or D3 are chosen, otherwise State D2 is used. When dialing is complete, the process signals whether the number is local or trunk. The Dialing process is subordinate to the Originating Switch process. It is started by the main process, and sends the result back to it. It thus illustrates how 2 processes, each exactly defined by a CSTD, can interact, one being subordinate to the other.

PROCEDURE 10 (b):

Dialing is specified by a separate CSTD. It is subordinate to the main CSTD of the Originating Switch. It functions when a call is in State S2 in that main diagram. Its function is to receive the dialled digits, identify local and trunk calls and return control to the main CSTD when a complete valid number is received. All States can return to *Idle*.

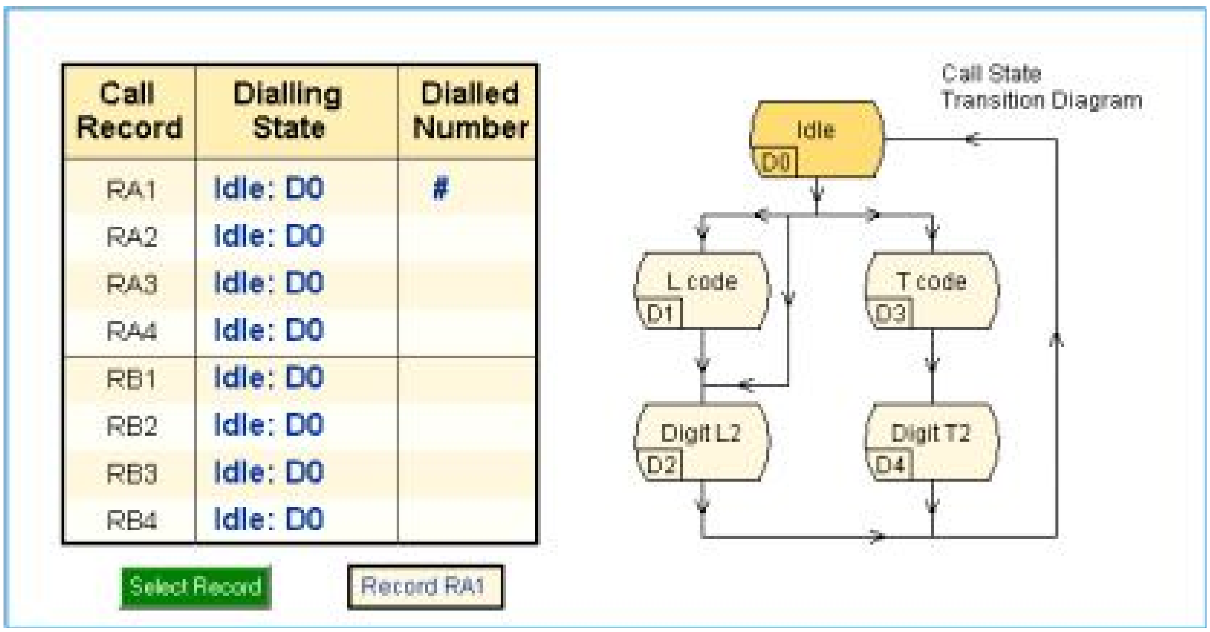

Figure 10.4 Dialing CSTD and Record

Use the telephones to make various local and trunk connections. See how 2 or 3 digit local numbers are handled.

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QUESTIONS:

1. How are local and trunk numbers distinguished from each other?

2. Why is State D1 sometimes used for local calls, and sometimes not?

3. What State is the main Originating CSTD at when the Dialing CSTD is active?

THEORY 10(c):

Destination Switch CSTD:

The Destination Switch CSTD defines the process for receiving trunk connections. It is much simpler than the Originating Switch CSTD. If the line is available, Ringing starts, and Ring Tone is returned to the trunk connection. If not, a suitable Message is returned to the Originating Switch, and the process is closed by returning to Idle.

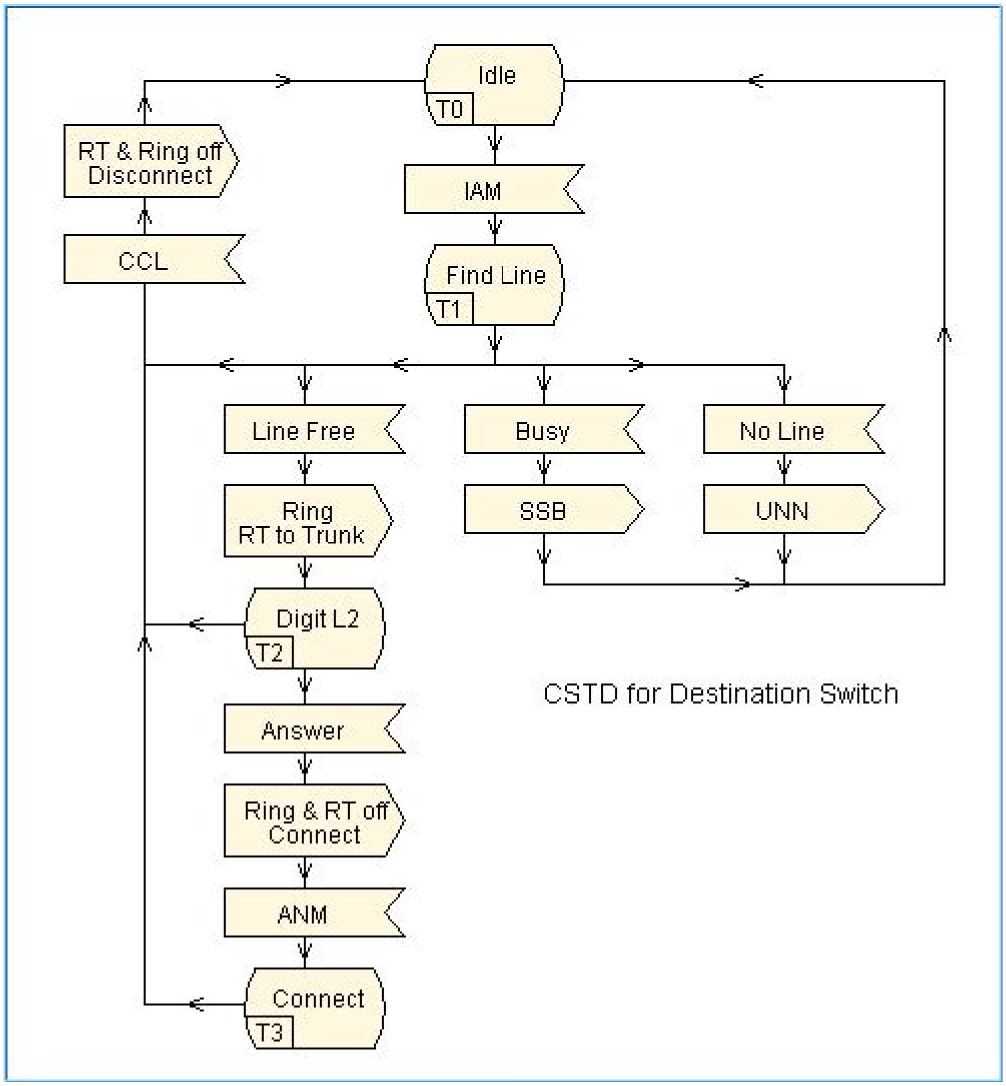

Figure 10.5 Destination CSTD

If the telephone at the required Line goes off hook, the answer message ANM is sent to the Originating Switch, the tones and ringing are cleared, and the lines connected. At any stage, if a forward clear message CCL is received, the line is cleared, and the process returns to Idle.

The Destination Switch CSTD must be designed to interact correctly with the Originating Switch CSTD. Neither of them is subordinate to the other, but they must work with each other. Both Switching Centres are able to act as Originating or Destination Switch; therefore both processes are available in each Switch.

PROCEDURE 10(c):

The operation of a trunk call at the Destination Switch is specified by an independent CSTD. When dialing for a trunk call is complete, a Message is sent to the Destination Switch with the local digits of the dialed number. If that line is available, then Ringing starts and Ring Tone is returned to the Originating Switch.

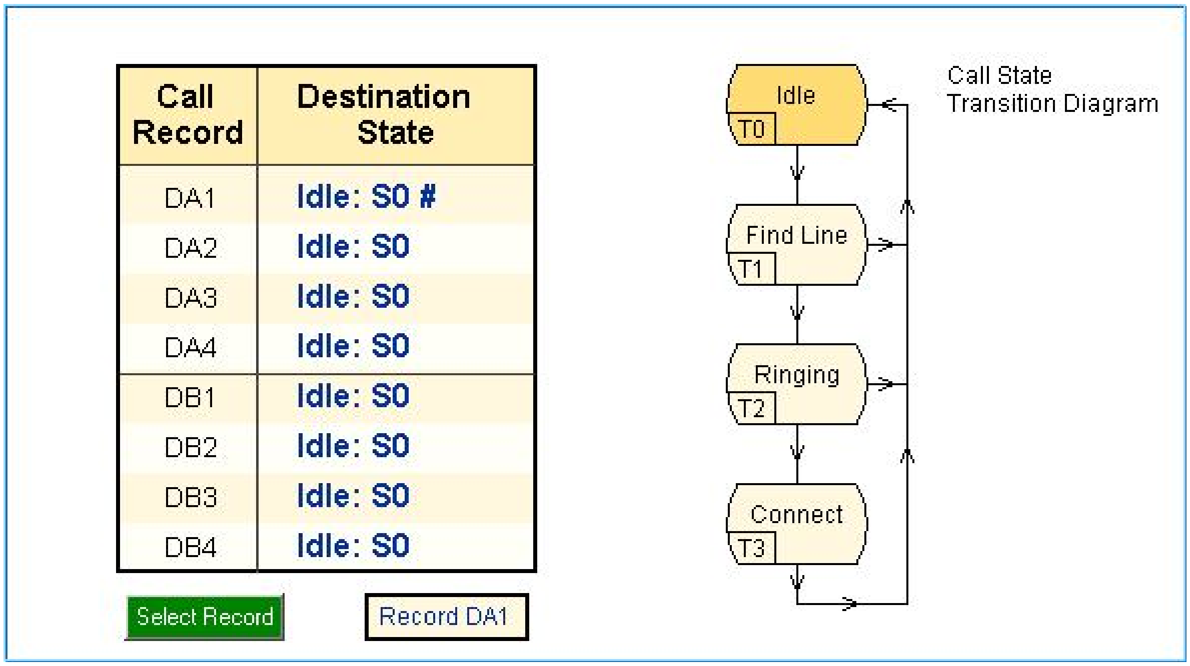

Figure 10.6 Destination CSTD and Record

If not a Message for Busy, or Number Unobtainable is sent to the Originating Switch. If the telephone goes off hook, an ANM message is sent to the Originating Switch. The process returns to Idle when a forward clear message CCL is received from the Originating Switch. The # sign shows which record is displayed. The ***Select Record*** button can be used to show any other record.

Use the telephones to make various trunk connections.

QUESTIONS:

1. At what stage of a call does the Destination CSTD start to operate?

2. What State is the main Originating CSTD in when the Destination CSTD is at

(i) State T2 - Ringing,

(ii) State T3 - Connect?

3. What happens at the Destination Switch if a call is cleared when the Destination CSTD is at State T2 - Ringing?

4. What circumstances would prevent the use of States T2 and T3?

LAB SESSION# 11

TRUNK CALL PROGRESS

OBJECTIVES:

- To analyze the function of point codes and circuit identification codes.
- To determine the mechanism for timing trunk calls.
- To evaluate the call progress in terms of states.
- To establish the characteristics of numbering schemes for multi-switch systems.

THEORY 11(a):

Signal Exchange Diagrams:

A useful method of understanding the progress of a call is a Signal Exchange Diagram. An alternative name for the diagram is a Signal Sequence Diagram. The Diagram displays the Signals exchanged between Lines and Switching Centres. Distance is displayed horizontally. The sequence develops downwards.

The first Diagram shows the course of a normal Call. It shows which Signals are exchanged between which points. It also shows the correct sequence of Signals in normal circumstances, and when each stage of connection or disconnection is made.

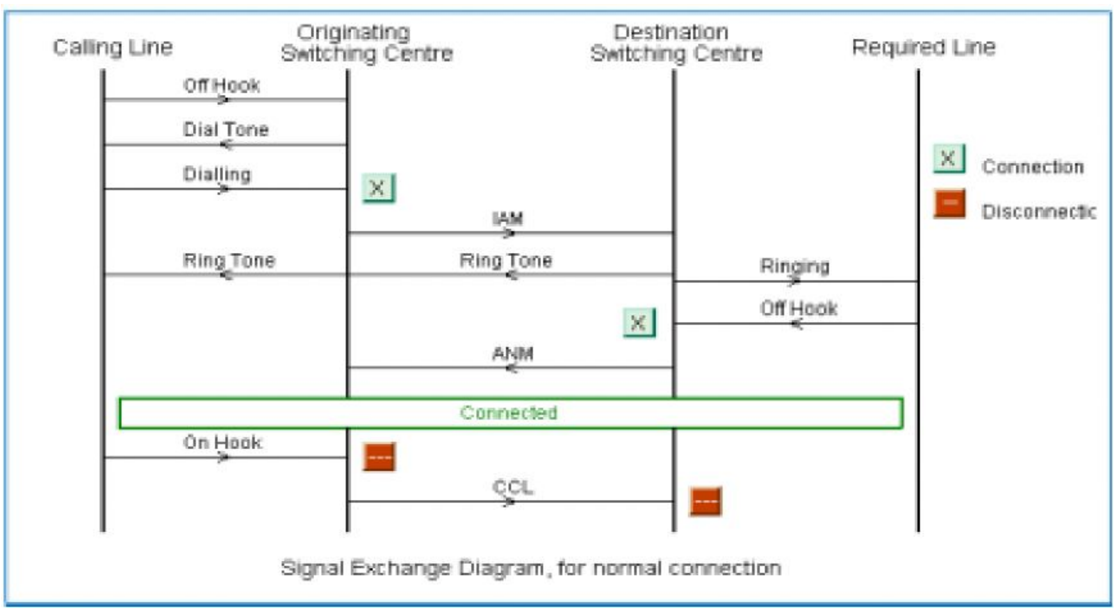

Figure 11.1 Signal Exchange Diagram

Unsuccessful Call Signals:

The other Signals used for unsuccessful connection attempts can also be shown.

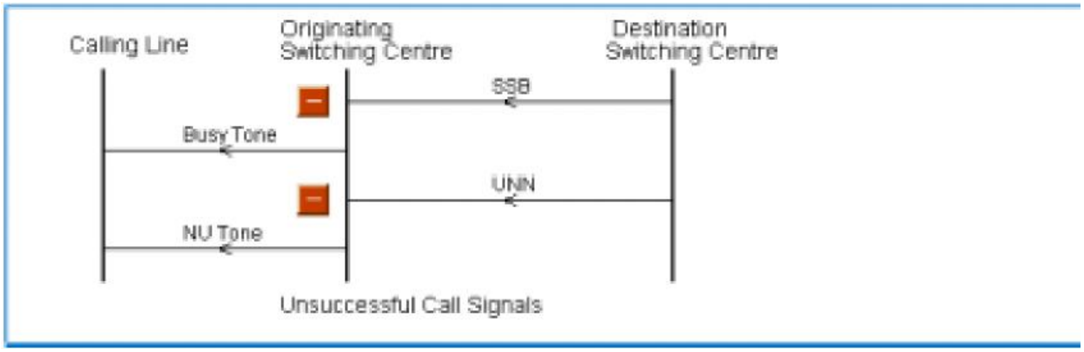

Figure 11.2 Unsuccessful Call Signals

These Signals could be included in the first diagram, and would follow the IAM Message.

Call Progress in Originating Switch:

Call Progress is observed by following the entries in the Call Record. The Call Records for both Switches, when acting as Originating Switches, are shown. Many of the items in the Call Record are similar to those in the Single Switch Assignments. However, there are extra items for trunk connections. The Point Code PC and the Circuit Identification Code CIC are determined and stored. They form the Label which is used for all Messages for any trunk connection.

For trunk connections the Required Line Identity RLI is not known. It is determined from the Directory Numbers used at the Destination Switch. Also for trunk connections there is no use for the Ring Count, because Ringing and Ring Tone are provided at the Destination Switch.

Charging information is kept at the Originating Switch. This is the reason for State S9. Only after the answer Message ANM has been received does the Call Duration refer to chargeable use. For local connections, there is no requirement for a Message Label. The Call Record is kept open until the Calling Line goes On Hook.

PROCEDURE 11(a):

Each connection is controlled by the Switching Centre at which the call originates. The Call Record has the necessary information for both local and trunk connections.

Call Record		Switching Centre A				Switching Centre B			
		RA1	RA2	RA3	RA4	RB1	RB2	RB3	RB4
Call State	CS	S0	S0	S0	S0	S0	S0	S0	S0
Calling Line Identity	CLI	---	---	---	---	---	---	---	---
Destination Address	DA	---	---	---	---	---	---	---	---
Req'd Line Identity	RLI	---	---	---	---	---	---	---	---
Point Code	PC	---	---	---	---	---	---	---	---
Circuit Identity Code	CIC	---	---	---	---	---	---	---	---
Call Duration	CD	---	---	---	---	---	---	---	---
Ring Count	RC	---	---	---	---	---	---	---	---

Figure 11.3 Call Records of Switching Centres

Make local calls and trunk calls and observe the various sequences.

Make trunk calls to Busy & NU numbers.

QUESTIONS:

1. Which kind of connection uses the Required Line Identity RLI, local or trunk? Why?

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2. What are the Point Code and Circuit Identification codes required for?

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3. Is the Ring Count used for trunk calls?

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4. When is the Call Duration restarted in a trunk call?

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THEORY 11(b):

Dialing:

Dialing is defined by its own CSTD, and thus has its own Call Record. Again the Call Records for both Switches are shown. The Destination Address *DA1* to *DA3* is obtained during the Dialing process. The first digit defines whether it is a local or trunk call. For a local call, the succeeding digits DA2 and DA3 are enough to define the Required Line Identity. For a trunk call all 3 digits are used. However, in this Assignment, there is only one other Switching Centre available!

The Record is closed when Dialing is complete, by returning to State D0.

PROCEDURE 11(b):

As seen in the CSTD Assignment, Dialing is an independent process, and therefore has its own Call Record. Each digit is stored in the appropriate Destination Address location DA1 to DA3. The Record is opened when the Originating CSTD reaches State 2. It is closed by setting the State to D0, immediately the dialing is complete, and the dialed number is passed to the main process.

		Switching Centre A				Switching Centre B			
Call Record		RA1	RA2	RA3	RA4	RB1	RB2	RB3	RB4
Dialling State	DS	D0	D0	D0	D0	D0	D0	D0	D0
Digit	DA1	---	---	---	---	---	---	---	---
Digit	DA2	---	---	---	---	---	---	---	---
Digit	DA3	---	---	---	---	---	---	---	---

Figure 11.4 Call Record for Dialing State

Use the telephones to make various local and trunk connections. See how 2 or 3 digit local calls are handled.

QUESTIONS:

1. Assume that Switching Centre A has Identity 7 and the Lines LA1 to LA4 are numbered 22, 33, 44 and 55 respectively and Switching Centre B has identity 6 and similar numbers for Lines LB1 to LB4.

2. What is the response if the following numbers are dialed:

(i) 622 at Switch A;

(ii) 633 at Switch B;

(iii) 721 at Switch A

THEORY 11(c):

Destination Switch:

The Call Records are shown for both Switches when acting as Destination Switches. Any trunk connection uses an Originating Call Record at one Switch and a Destination Call Record at the other Switch. The Destination Switch records the information received from the Initial Address Message. The Label information, PC and CIC, is kept to provide the Label for Messages back to the Originating Switch. The Destination Address is used to determine the status of the Line indicated. If the Address corresponds to a Line, then the Required Line Identity is determined. If the Line is free, then Ringing commences, and Ring Tone is returned to the trunk connection, to be transmitted to the Calling Line. The Record is kept open until a Calling Party Clear CCL Message is received from the Originating switch. This may occur before or after the Line has been answered. If the Line is busy, or Unallocated, Messages are sent to that effect to the Originating Switch, and the Record is closed. If there is no free line, the Record is closed quickly. Therefore the first State T1 is held for observation in all cases.

PROCEDURE 11(c):

Each Switching Centre has to accept incoming trunk calls. These require a separate set of Call Records at each Switch.

		Switching Centre A				Switching Centre B			
Call Record		RA1	RA2	RA3	RA4	RB1	RB2	RB3	RB4
Destination Trunk State	DTS	T0	T0	T0	T0	T0	T0	T0	T0
Point Code	PC	---	---	---	---	---	---	---	---
Circuit Identity Code	CIC	---	---	---	---	---	---	---	---
Destination Address	DA	---	---	---	---	---	---	---	---
Dest Req'd Line Identity	DRLI	---	---	---	---	---	---	---	---
Destination Ring Count	RC	---	---	---	---	---	---	---	---

Figure 11.5 Call Record for Destination State

Use the telephones to make various trunk connections. Record your observations.

QUESTIONS:

1. What items in the Destination Call Record are derived from the Initial Address Message?

2. Why are the Point Codes and Circuit Identification Codes retained?

3. Why is Call Duration not recorded at the Destination Switch?

ASSIGNMENTS WITH TRUNK NETWORKS BOARD

LAB SESSION# 12

TRANSIT SWITCHING CENTRE

OBJECTIVES:

- To investigate the numbering schemes applied to multi-switching centre telephony systems.
- To establish the need for a system using transit switching centres.
- To identify the components of the control messages transmitted between centres.

Preliminary Procedure:

Check that your Workstation is set up with two Digital Switch Centres and Telephone Trays (58-1 22 and 58-1 23) and one Trunk Networks Board (58-140). One Switching Centre Type should be set to ‘A’ (green LED) and the other to ‘B’ (yellow LED). A ‘curly’ trunk cable should interconnect the A ‘Trunks’ connector to ‘Trunk A’ on the 58-140. Similarly, another ‘curly’ trunk cable should interconnect the B ‘Trunks’ connector to ‘Trunk B’ on the 58-140. All four ‘Bus Select’ switches should be set to ‘2’ (green LED), and the ‘Level’ switch on the 58-140 should be set to ‘Single’

THEORY 12(a):

Transit Switching Centres:

The basic functions of Transit Switching Centres are simpler than those of a Local Switch. There are no line circuits, which provide the BORSCHT functions, including analogue to digital conversion, for every telephone; and there is no call accounting. All inlets and outlets are purely digital.

However, they may be needed to accept a large number of connections. A typical large Digital Switch may have a capacity of up to 100,000 lines. To do this many different combinations of time and space switching are used. They also have a significant control function, including signaling, routing and switching. A large Switch may accept up to 500,000 call attempts during the busy hour.

Long Distance Transmission:

The distances between Transit Switches are naturally greater than those between Local Switches and the subscribers. Hence long distance transmission is required. Digital signals can be transmitted over the same twisted pairs or co-axial cables used for analogue signals. Regenerative Receivers are used at frequent intervals to restore the signals before attenuation and noise have rendered them unrecognizable. This ability to restore signals exactly is one of the major advantages of digital systems. A typical distance between repeaters is about 2000 m. Microwave radio links, operating at frequencies between 2 and 13 GHz, are used, but are limited to line of sight spacing of about 50 km.

A special case of radio is the use of satellites. For a geostationary satellite there is a propagation delay of about 0.25 seconds in each direction. This is noticeable, but tolerable, during conversation.

Thus CEPT traffic using 2.048 Mbps or multiples of that basic rate, T1 traffic at multiples of 1.544 Mbps, and other signals can all be transmitted along the same link, and each can be accessed without complete demodulation. A transmission system developed rapidly in the mid 1990s is Asynchronous Transfer Mode ATM, otherwise known as Broadband ISDN, where ISDN is the Integrated Services Digital Network.

ATM uses cells of 53 bytes, 5 for a header and 48 for user data. The initial data rates are 155.52 and 622.08 Mbps, corresponding to the SONET rates.

Synchronization:

An important requirement of long distance digital transmission is the need for the digital clocks at each Switching Centre to be consistent. The different clocks may have slightly different frequencies, and they may fluctuate with time.

The clocks in a particular region can be controlled by a master clock, so that they are all kept synchronous.

At boundaries between regions it may be necessary to lose or insert bits. This is serious as it can disrupt the frame structure of synchronous systems, and must be carefully controlled. There is a delay in propagation along transmission links, which may also fluctuate, causing jitter. Errors due to jitter are overcome by elastic stores, which hold the data and re-synchronize it with the receiving clock.

Numbering:

Structure with Trunk Networks Workboard:

Two Digital Switching Centre Workboards are connected to the Trunk Networks Workboard by Trunk Cords; Workboard A to the socket ***Trunk A*** and Workboard B to socket ***Trunk B.*** For all Assignments using the Trunk Networks Workboard, the telephones are organized as if each Switching Centre Workboard actually had two Switching Centres, each with 2 telephones. This provides more possibilities for demonstrating the switching procedures.

The two Switching Centres on Workboard type A are designated ***C*** and ***D,*** and those on Workboard type B are ***E*** and ***F.*** The numbering for each Assignment is preset according to its particular requirements, and cannot be altered.

Bus Select settings:

The Inlet Bus Select and Outlet Bus Select Switches must both be in position *Bus 2* for the Assignments using the Trunk Networks Workboards. If this is not done, the Practicals will operate, but there will be no speech path between the telephones.

Level Switch:

For this and the next Assignment, the Level Switch on the Trunk Networks Workboard must be in the *Single Level* position.

Numbering and Tones:

For this Assignment all of the 4 Local Switching Centres are connected to one Transit Switching Centre. This Switch is designated ***X.*** Each Local Switch, ***C*** to ***F,*** has a unique Directory Number, from 4 to 7. The Directory Numbers for the telephones at each Local Switch are deliberately identical, 21 and 32. Three digit dialing is required for all calls, whether local or trunk.

The tones are those selected in the Trunk Configuration Assignment in the Dual Digital Switching Workboards section, either preset or programmed. Local Switches ***C*** and ***D*** use the tones set for Workboard A, and Switches ***E*** and ***F*** use those for Workboard B.

This Practical is designed to familiarize the user with the numbering system and to consider its implications.

PROCEDURE 12(a):

The first Assignments on a Trunk Network assume that 4 Local Switching Centres are all connected to one Transit Switching Centre. For simplicity the line numbers are not programmable in these Trunk Networks Assignments.

Lines 1 and 2 on the Digital Switching Centre set to *A* are assumed to be 2 lines in Local Switch ***C.*** Lines 3 and 4 are the 2 lines on Local Switch ***D.*** Similarly Switching Centre *B* provides Local Switches ***E*** and ***F.*** All Bus Select Switches MUST be in position 2 to provide speech paths.

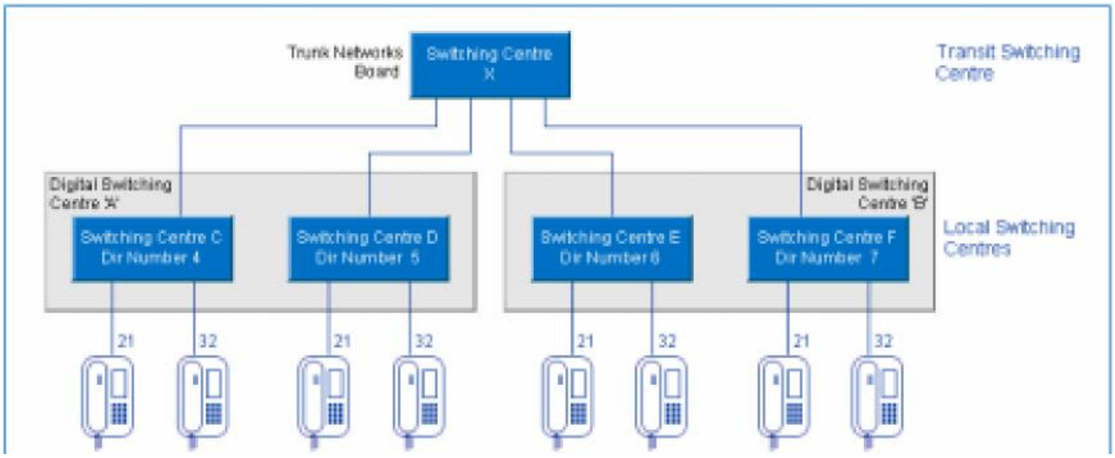

Figure 12.1 Trunked Network Architecture

Any Local Switch may use any directory numbers. To demonstrate this, the same two numbers (21 and 32) are used for the telephones in all Local Switches. Each Local Switch has its own directory number, from 4 to 7.

Make local and trunk calls and determine how many digits are needed for each.

QUESTIONS:

1. How many digits are required for dialing local and trunk calls?

2. How many telephones could be connected to each Local Switch, using this numbering scheme; and how many Switches could be accommodated?

3. Hence what is the maximum number of telephones which could be accommodated with this 3 digit numbering scheme?

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THEORY 12(b):

Transit Switch CSTD:

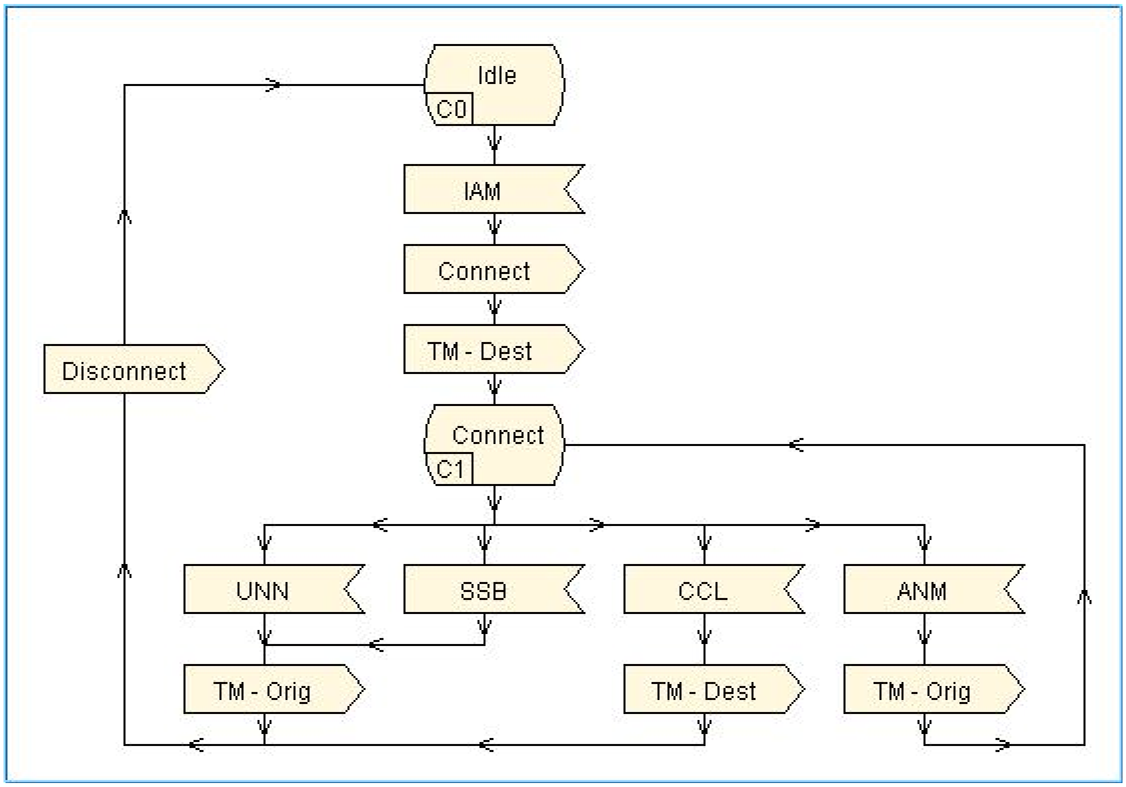

Figure 12.2 Transit Switch CSTD

The CSTD for the Transit Switch Centre only has the 2 states, **Idle** and **Connect.** TM is Transmit Message to either Originating (Orig) or Destination (Dest) Switching Centre.

Transit Switch Operation:

When an Initial Address Message (IAM) is received the following *Tasks* are undertaken:

1. A Call Record is opened.
2. The route for the call is found by identifying the required Local Switch. The *Outputs* are:
3. The circuit is connected in both directions.
4. The IAM is transmitted to the Destination Switch.

A *Transition* is made to the Connection State C1, *Connect.*

- After that any messages which are received from the Destination Switch are examined and transmitted to the Originating Switch.
- Similarly messages from the Originating Switch are examined, and transmitted to the Destination.
- If a Calling Party Clear (CCL) message from the Originating Switch, or Busy (SSB) or Unknown Number (UNN) from the Destination are received, then the:
- *Task* is to disconnect the circuit
- *Transition* is made back to State C0, *Idle.*

PROCEDURE 12(b):

The CSTD for a Transit Switching Centre is very simple. The functions of the Transit Switch are to select a route for a call and to connect or disconnect the circuit according to the messages it receives; and to transmit suitable messages to the Originating or Destination Switching Centres.

If a circuit has been connected by the Transit Switch, then the CSTD is in the *Connect State,* else it is *Idle.*

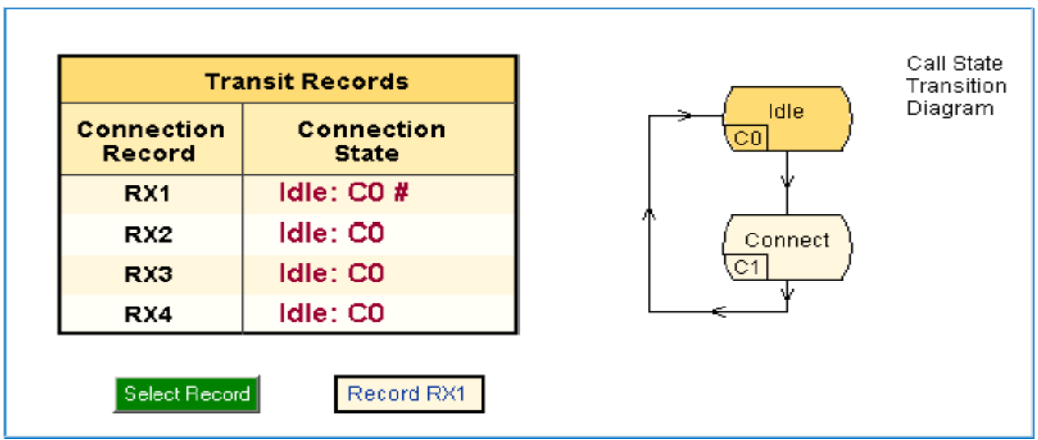

Figure 12.3 Transit Switch CSTD

The **Call Record** menu can be used to obtain the CSTD for different calls.

Use the telephones to make various trunk connections.

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QUESTIONS:

1. What causes a Transition from *Idle* to *Connect* at a Transit Switch?

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2. What can cause a Transition back to *Idle?*

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3. What Outputs and Tasks occur if a Busy (SSB) message is received from the Destination Switch?

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THEORY 12(c):

Transit Signaling:

Messages between Transit Switching Centres:

In order to establish a trunk connection through a Transit Switching Centre, control Messages are sent from the Originating Local Switch to the Transit Switch; and thence to the Destination Switch. Connections also have to be made over those two links. The Label used for Messages over each link are made up of a Point Code (PC), derived from the identity of the Switches at the end of each link, and a Circuit Identification Code (CIC), specifying the trunk and timeslot used. Therefore the Labels are different for the Messages on each link.

For each link, the most recent Message is shown in abbreviated form in the Practical. Each Message can also be shown in full by clicking on the short message legend. Remembering that the Least Significant Bit is transmitted first, the sequence is Length Indicator, Label (2 bytes), Heading Code, and Destination Address where necessary.

Message Contents:

The sequence of Messages for any connection is identical to that for the Dual Switching Centre Assignments.

The Destination Address, sent with the Initial Address Message IAM, is different at each stage. All 3 dialed digits are sent from the Local Switching Centre to the Transit Switch. When the Transit Switch has identified the required Local Switch, it is not necessary to send the first digit any further. Thus only 2 digits are sent on to the Local Switch.

A number can be dialed for which there is no Local Switching Centre. This is recognised by the Trunk Switching Centre, which immediately returns the Unallocated Number UNN message.

The other Messages used are again Answer ANM, Calling Party Clear CCL and Subscriber Busy SSB.

PROCEDURE 12(c):

Trunk calls between the 4 local Switching Centres require 2 interconnections between Switches.

The control messages follow the same route, and therefore there are two stages for each message, into the Transit Switch and then to the required Local Switch.

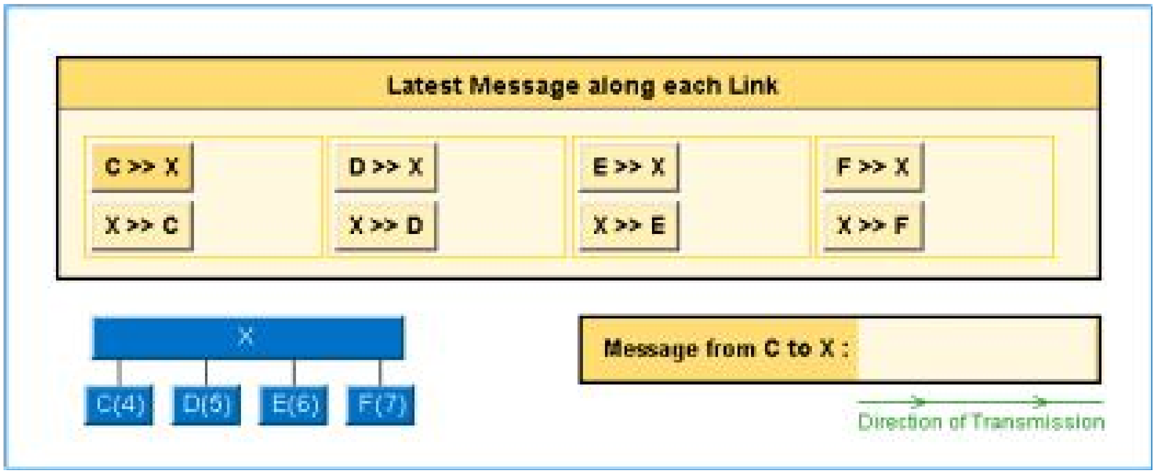

Figure 12.4 Transit Signaling

The latest message along each link is shown in abbreviated form.

Each message can be displayed in full by clicking on the legend by each short message.

Use the telephones to make various trunk connections and observe the sequence of messages.

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QUESTIONS

1. What is the sequence of Messages for a successful connection from telephone 521 to 632, which is answered and then cleared?

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2. Why is the number of digits sent with the IAM different for the 2 stages?

3. If the number 821 is dialed, what is the reaction of the system?

LAB SESSION# 13

TRANSIT SWITCHING CALL PROGRESS

OBJECTIVES:

- To investigate the structure of the control message associated with the transit switching centre.
- To analyze the use of the codes contained within the message.
- To determine the route through the switch.
- To establish the functions of the switch.

THEORY 13(a):

Switching Centre Control:

Any telephone Switching Centre has 3 major functions for controlling a connection: signaling, routing and switching. There are also other functions such as testing, and call accounting. Hence there is a great use of computers and microprocessors of different kinds for the different functions; and they have different time requirements. Setting up and supervising calls must be performed as quickly as possible. Logging the call accounting information must be reasonably prompt, but not as fast as call processing. Other functions such as testing or the management of databases of information about the network have much less urgency.

Some of the high speed functions are typically performed by processes written in assembly language. As much as possible is written in high level languages, for ease of production and to assist in maintenance or further development of the system. A specific language for controlling telephone switching is the CCITT (now ITU-T) High Level Language CHILL, but many others have been used.

It has been estimated that possibly 5% of code is written in assembly language, but that part might be used for 95% of the operations!

Operation of Digital Switching Centre:

The Digital Switching Centre Workboard is not a typical Switch, because it is used to demonstrate particular features for educational reasons. However the control structure illustrates some aspects of more commercial systems. The Switching Centres are run by a microprocessor in the Controller. There is a communication system between the Discovery software in the computer and the Controller, operated by an EPROM in the Controller. The communication system is used to down-load the assembly language programme when necessary. The largest programme, for the Trunk Networks Assignments, is about 6500 bytes of code. The Controller programme structure is based on a 20 ms interrupt from hardware. There are 3 main sections to the code.

- **Timing** maintains all the clocks, such as ringing and call duration, which are based on 0.1 second units.

- **Line Scan** takes in the Switch Hook states, and the DTMF receiver outputs, and processes the calls appropriately, including sending messages to other Switching Centres. This is an extensive set of routines. The design of the call processing is based on the use of ITU-T Call State Transition Diagrams as covered in the Assignments.
- **Trunk Messages** reads the messages between the different Switching Centres and again processes the calls.

There are also routines for initialization, continuous testing and debugging. Thus the telephone system works essentially independently of the Discovery programs. Information about the system, as needed for each Practical, is acquired using the communication system.

Data Storage:

Although only one microprocessor is used, the data for each Switching Centre is held separately, and Messages are used to exchange Information. Hence the Switching Centres are genuinely independent of each other. Information about call accounting, and about the programmed tone system, is held in files in the computer, and so can be accessed whenever the Assignments are run.

Directory numbers and Bus Select data are held in protected areas in the Controller memory, and are maintained while it is switched on. They are used for all relevant Assignments.

Transit Switching Centre:

Call Progress:

The CSTD for a Transit Switching Centre has shown that the operations required are simpler than for the Local Switches. There are only 2 States, *Idle* and *Connect.* Hence the Call Records are simpler. There are several functions required in the Transit Switch.

- One function is to receive and transmit Messages between the Originating and Destination Switches. For this the Labels for inlet and outlet Messages are recorded. Each Label consists of the Point Code PC and the Circuit Identification Code CIC.

The Inlet Labels are defined by the Originating Switch, and received by the Transit Switch.

- Another function is to establish the route for the connection. In this case that consists of choosing which Local Switch is the destination of the connection. That is determined by the first digit of the Destination Address DA.

Then the Outlet Label is determined by the route. The OPC depends on the identity of the next Switch; and the OCIC describes the bus and timeslot to be used.

- The final function is switching; connecting the Inlet bus and timeslot to the Outlet bus and timeslot.

Holding a State for Observation:

If unsuccessful call attempts are made, to Busy lines or to Unallocated Numbers, in normal use the Call Record is cleared very quickly. Occasionally some entries are briefly visible on the screen.

The option is available of making Calls hold at the Destination State T1, so that the Call Records can be observed. The option can be turned on or off by the *State Hold* toggle button.

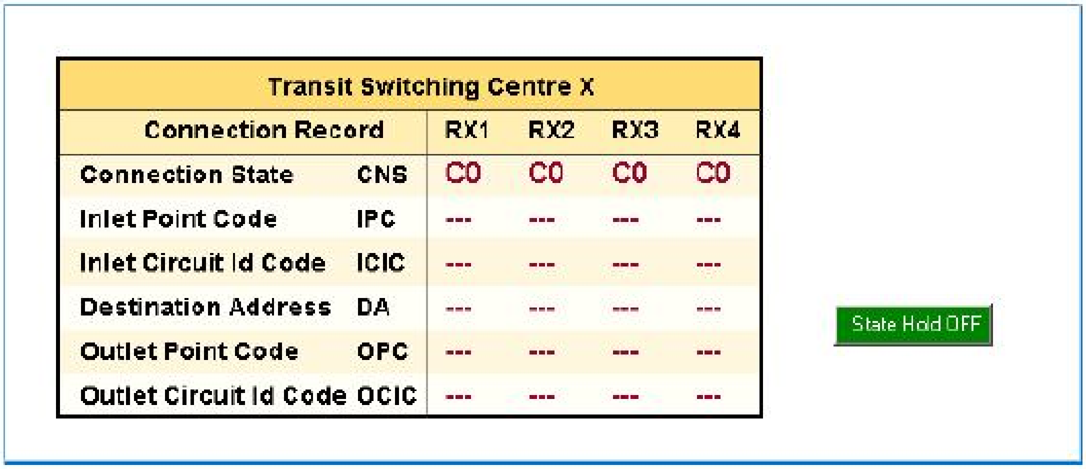
Level Switch:

The Level Switch should be in the *Single Level* position for this Assignment.

PROCEDURE 13(a):

The operation of a Transit Switching Centre requires data for making the connection and for recognizing and routing the control messages. The Circuit Identification Codes (CIC) contain the connection data. The labels (CIC plus PC) identify all messages in both directions.

The Call Records for unsuccessful call attempts are cleared very quickly. An option of holding calls to observe the record is provided by the *State Hold* toggle button.



Transit Switching Centre X					
Connection Record		RX1	RX2	RX3	RX4
Connection State	CNS	C0	C0	C0	C0
Inlet Point Code	IPC	---	---	---	---
Inlet Circuit Id Code	ICIC	---	---	---	---
Destination Address	DA	---	---	---	---
Outlet Point Code	OPC	---	---	---	---
Outlet Circuit Id Code	OCIC	---	---	---	---

Figure 13.1 Transit Switching Centre Record

Make trunk calls and observe the various sequences.

Make trunk calls to Busy & NU numbers. Examine the call records in the Originating Switch and Destination Switch practicals for the same call. Also consider the signals exchanged between the Switches, shown in the Transit Signaling practical.

QUESTIONS:

1. In the Transit Switch, what use is made of the destination Address DA?

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2. Where are the Inlet Labels (IPC and ICIC) and the Outlet Labels (OPC and OCIC) derived from?

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3. What are the 3 functions of the Transit Switch?

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THEORY 13(b):

Originating Switch Operation:

The operation of an Originating Switch is similar whatever trunk network it is connected to. Hence the Call Record is similar to that for the Dual Switching Centre Assignments. Now there are 4 Local Switching Centres to consider. There is a Call Record for each Switching Centre. Since there are only 2 telephones for each Local Switch, only 2 Call Records are needed for each Switch.

Labels:

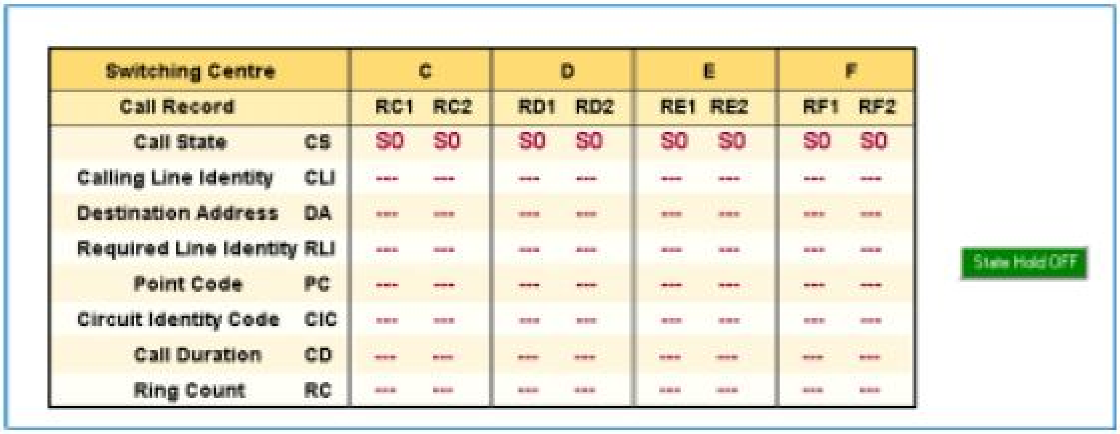
If a **Busy (SSB)** or **Unallocated Number (UNN)** Message is received, the Label is removed from the Call Record immediately. This is necessary because the connection to the Transit Switching Centre is cleared immediately. The timeslot is therefore available for another connection, while the tone is being heard by the original caller. A second call would use the same timeslot on the same trunk bus as the previous one.

The Circuit Identification Code CIC for each connection has the 3 most significant bits for the bus being used, and the remaining 5 bits for the timeslot. The second call would therefore have the same CIC as the previous unsuccessful one. Confusion would arise if 2 connections had the same CIC and hence the same Label.

Therefore the Label must be removed as soon as the SSB or UNN Message is received. By using the *State Hold* toggle button, the Label can be observed.

PROCEDURE 13(b):

The 4 Local Switching Centres C to F each have Call Records.



Switching Centre		C		D		E		F	
Call Record		RC1	RC2	RD1	RD2	RE1	RE2	RF1	RF2
Call State	CS	S0	S0	S0	S0	S0	S0	S0	S0
Calling Line Identity	CLI	---	---	---	---	---	---	---	---
Destination Address	DA	---	---	---	---	---	---	---	---
Required Line Identity	RLI	---	---	---	---	---	---	---	---
Point Code	PC	---	---	---	---	---	---	---	---
Circuit Identity Code	CIC	---	---	---	---	---	---	---	---
Call Duration	CD	---	---	---	---	---	---	---	---
Ring Count	RC	---	---	---	---	---	---	---	---

Figure 13.2 Switching Centre Call Record

Use the telephones to make various local and trunk connections. The *State Hold* menu can be used to observe the operation of unsuccessful call attempts.

QUESTIONS:

1. If a call is made from a telephone on Switching Centre C, what is a typical CIC?

2. What bus and what timeslot does this define?

3. What busses and what typical timeslots are used for Outlet connections from the other Local Switches?

THEORY 13(c):

Destination Switch

Again this is similar to the Destination Call Records for the Dual Switching Centre. There are 4 Local Switching Centres, and hence 4 Call Records are needed. In each Switch, there are only 2 telephones, and thus only two connections are possible. However, another unsuccessful attempt may be made after both lines are connected. Hence 3 Call Records are necessary. The *State Hold* toggle button is provided to enable the Call Record to be observed in the case of unsuccessful calls.

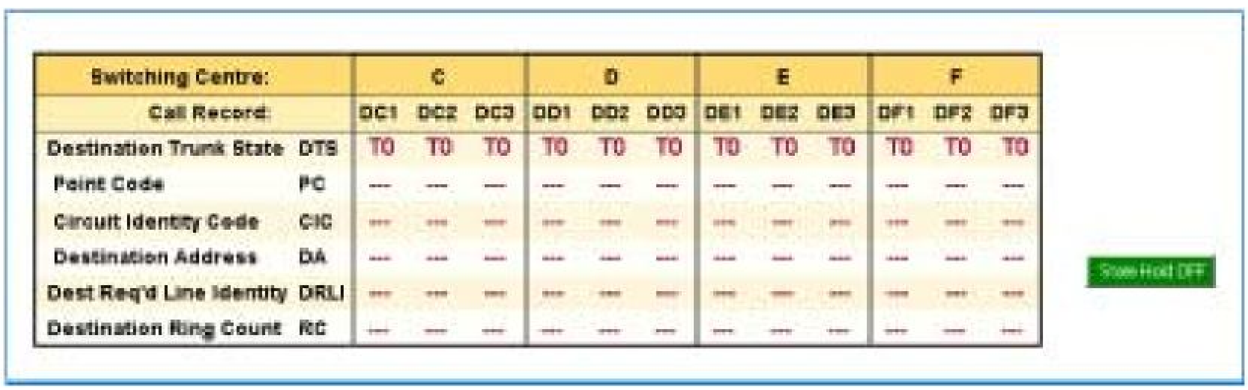
Bus identities:

The connection between each Switching Centre is a bus which has a different identity at each Switch; ie, at each end of the bus. For example, the bus between Switch ***C*** and Switch ***X*** is defined as 2 at ***C,*** and as 0 at ***X.*** Each Inlet and Outlet bus has the same identity at any particular Switch. For example, if an Inlet to Switch ***X*** is identified as 0, then the Outlet for the same bus is also 0. The identities are printed on the Workboard.

Different groups of timeslots are reserved on each bus for connections originating in each direction. They all provide 2 way connections. The bus identity and timeslot number are defined in the Circuit Identification Code CIC for each link in the connection.

PROCEDURE13(c):

Each Switching Centre has to accept incoming trunk calls. These require a separate set of Call Records at each Switch.



Switching Centre:		C			D			E			F		
Call Record:		DC1	DC2	DC3	DD1	DD2	DD3	DE1	DE2	DE3	DF1	DF2	DF3
Destination Trunk State	DTS	T0	T0	T0	T0	T0	T0	T0	T0	T0	T0	T0	T0
Point Code	PC	---	---	---	---	---	---	---	---	---	---	---	---
Circuit Identity Code	CIC	---	---	---	---	---	---	---	---	---	---	---	---
Destination Address	DA	---	---	---	---	---	---	---	---	---	---	---	---
Dest Req'd Line Identity	DRLI	---	---	---	---	---	---	---	---	---	---	---	---
Destination Ring Count	RC	---	---	---	---	---	---	---	---	---	---	---	---

Figure 13.3 Switching Centre Call Record

Use the telephones to make various trunk connections. The *State Hold* toggle button can be used to observe the operation of unsuccessful call attempts.

QUESTIONS:

1. Why are 3 Call Records required for each Local Switch when there are only 2 telephones at each one?

2. What are the identities of the Inlet busses of each Local Switch?

3. Inlet and Outlet connections use the same bus. Why are different identities used for connections originating in different directions?

LAB ASSIGNMENT

BASIC VoIP NETWORK CONFIGURATION

OBJECTIVES:

- To understand the working and configuration procedure of an IP Phone.
- To understand the procedure required to connect an ordinary Analog Phone to the Packet Switched Network
- To simulate a basic VoIP Network Configuration using Packet Tracer 5.3

PRE-REQUISITES:

Download and Install Packet Tracer 5.3 version on your computer

THEORY:

Cisco 2800 Series Integrated Services Routers:

Cisco 2800 Series Integrated Services Routers comprise four models: Cisco 2801, Cisco 2811, Cisco 2821, and Cisco 2851 routers. The 2800 Series routers provide up to 5 times the overall performance, up to 10 times the security and voice performance, embedded service options, and dramatically increased slot performance and density. The series also maintains support for most of the more than 90 modules that are available for the Cisco 1700 Series Modular Access Routers, 2600 Series Multiservice Platforms, and 3700 Series Multiservice Access Routers. The 2800 Series routers can deliver simultaneous, high-quality, wire-speed services up to multiple T1/E1or xDSL connections. The routers offer embedded encryption acceleration and, on the motherboard, voice digital signal processor (DSP) slots. They also offer intrusion prevention system (IPS) and firewall functions; optional integrated call processing and voicemail support; high-density interfaces for a wide range of wired and wireless connectivity requirements; and sufficient performance and slot density for future network expansion requirements and advanced applications.

Cisco 7960 IP Phone:

Power up the phone:

Two options are available in packet tracer for powering up the 7960 IP Phone:

- External power adapter
- PoE (only with 3560 multilayer switch)

If you choose to use the external power adapter, go to the physical tab and drag and drop the "IP_PHONE_POWER_ADAPTER" to the bottom left connector of the 7960 IP Phone.

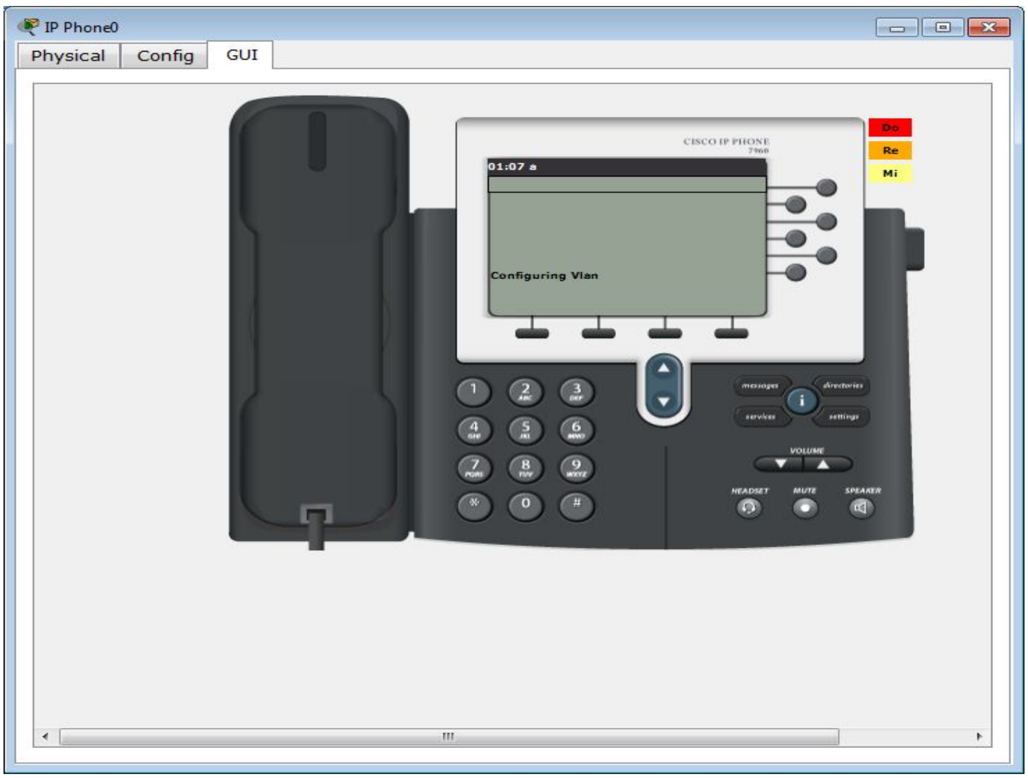


Figure 14.1 IP Phone GUI (Packet Tracer 5.3)

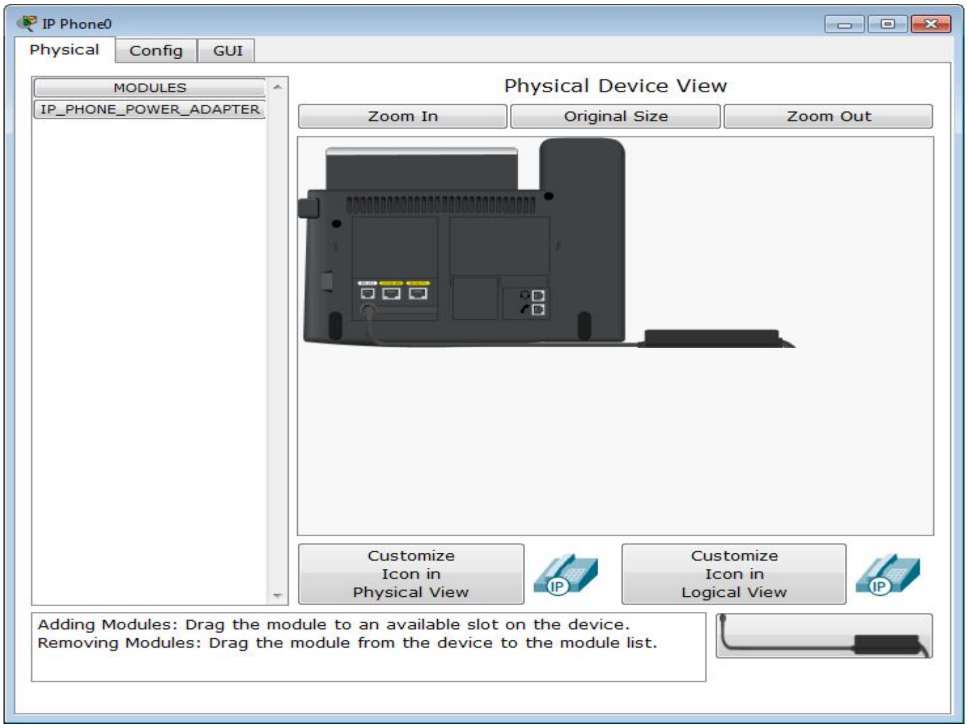


Figure 14.2 IP Phone Physical Interface (Packet Tracer 5.3)

If you want to use the PoE functionality of the Cisco 3560 switch, apply the following configuration to the switch interface connected to the phone:

Switch(config)#int fastEthernet 0/1

Switch(config-if)#power inline auto

Place calls:

The 7960 IP Phone does not have any configurable options. It receives it's IP address through DHCP and it's line number from the Call Manager Express server.

In the GUI tab, you can place a call, answer a call, and send Do, Re, and Mi notes to the recipient phone. To place a call, enter the recipient's line number first using the keypad and then click on the handset to dial out. To answer a phone call on the analog phone, click on the handset when the phone is ringing. While the line is connected, you can send Do, Re, or Mi to the recipient by pressing the respective buttons. In order to hear the sounds, be sure Sound is enabled in Preferences. To end a call, click on the handset.

Home VOIP device:

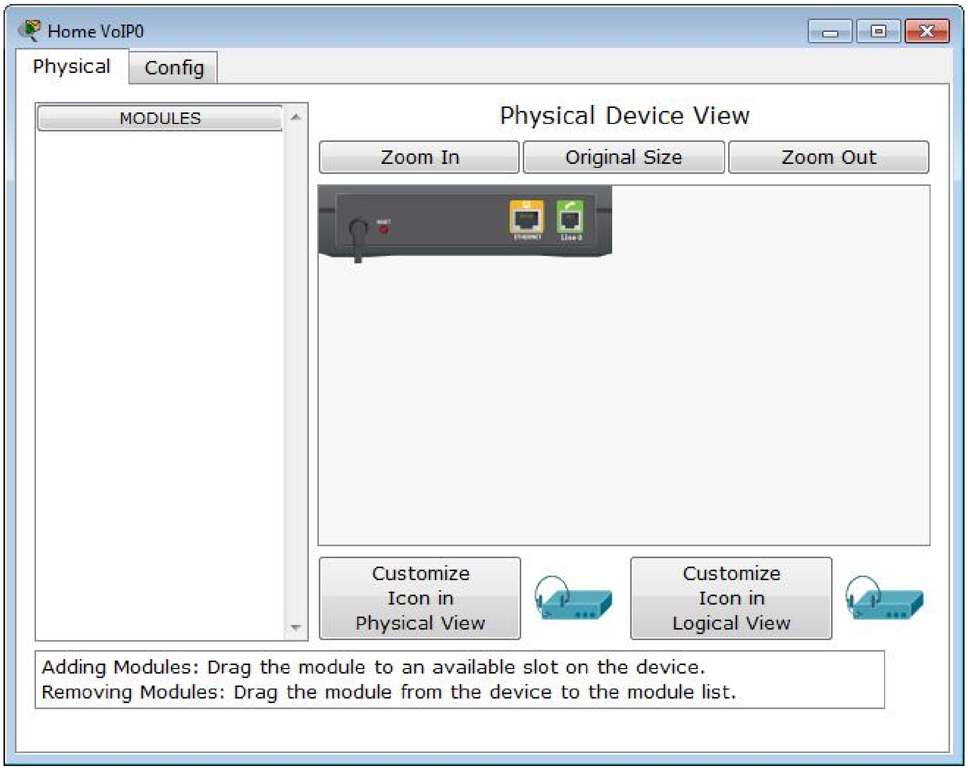


Figure 14.3 Home VoIP Device Physical Interface (Packet Tracer 5.3)

The Home VoIP only has a "Server Address" configuration in which you have to place the Call Manager Express IP address.

PROCEDURE:

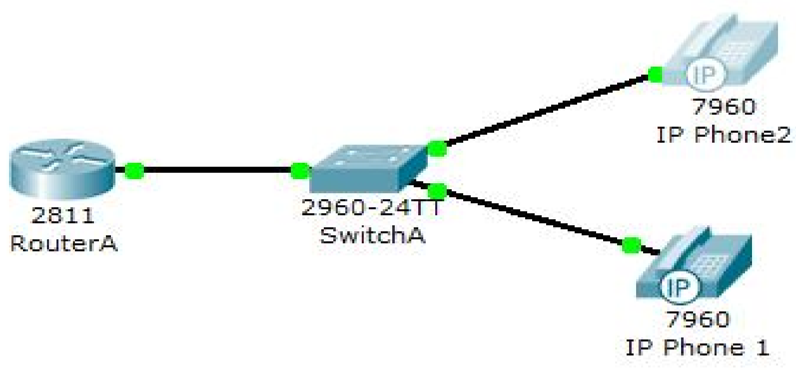


Figure 14.4 Topology to be implemented using Packet Tracer 5.3

Task 1: Topology

- Use Packet tracer 5.3 GUI to drag and drop the devices required to implement topology shown in figure 14.4.
- Connect the devices using appropriate cabling.

Task 2: Configure interface of router A connecting to Switch A and DHCP server on RouterA (2811 router)

Configure the FA 0/0 interface

RouterA>enable

RouterA#configure terminal

RouterA(config)#interface FastEthernet0/0

RouterA(config-if)#ip address 192.168.10.1 255.255.255.0

RouterA(config-if)#no shutdown

The DHCP server is needed to provide an IP address and the TFTP server location for each IP phone connected to the network.

RouterA(config)#ip dhcp pool VOICE #Create DHCP pool named VOICE

RouterA(dhcp-config)#network 192.168.10.0 255.255.255.0 #DHCP network 192.168.10 with /24 mask#

RouterA(dhcp-config)#default-router 192.168.10.1 #The default router IP address#

RouterA(dhcp-config)#option 150 ip 192.168.10.1 #Mandatory for VoIP configuration.

After the configuration, wait a moment and check that 'IP Phone 1' has received an IP address by placing your cursor over the phone until a configuration summary appears.

Task 3: Configure the Call Manager Express telephony service on RouterA

You must now configure the Call Manager Express telephony service on RouterA to enable VoIP on your network.

RouterA(config)#telephony-service #Configuring the router for telephony services#

RouterA(config-telephony)#max-dn 5 #Define the maximum number of directory numbers#

RouterA(config-telephony)#max-ephones 5 #Define the maximum number of phones#

RouterA(config-telephony)#ip source-address 192.168.10.1 port 2000 #IP Address source#

RouterA(config-telephony)#auto assign 4 to 6 #Automatically assigning ext numbers to buttons#

RouterA(config-telephony)#auto assign 1 to 5 #Automatically assigning ext numbers to buttons#

Task 4: Configure a voice VLAN on SwitchA

Apply the following configuration on SwitchA interfaces. This configuration will separate voice and data traffic in different VLANs on SwitchA. Data packets will be carried on the access VLAN.

SwitchA(config)#interface range fa0/1 – 5 #Configure interface range#

SwitchA(config-if-range)#switchport mode access

SwitchA(config-if-range)#switchport voice vlan 1 #Define the VLAN on which voice packets will be handled#

Task 5: Configure the phone directory for IP Phone 1

Although 'IP Phone 1' is already connected to SwitchA, it needs additional configuration before being able to communicate. You need to configure RouterA CME to assign a phone number to this IP phone.

RouterA(config)#ephone-dn 1 #Defining the first directory entry#

RouterA(config-ephone-dn)#number 54001 #Assign the phone number to this entry#

Task 6: Verify the configuration

Ensure that the IP Phone receives an IP Address and the phone number 54001 from RouterA (this can take a short while).

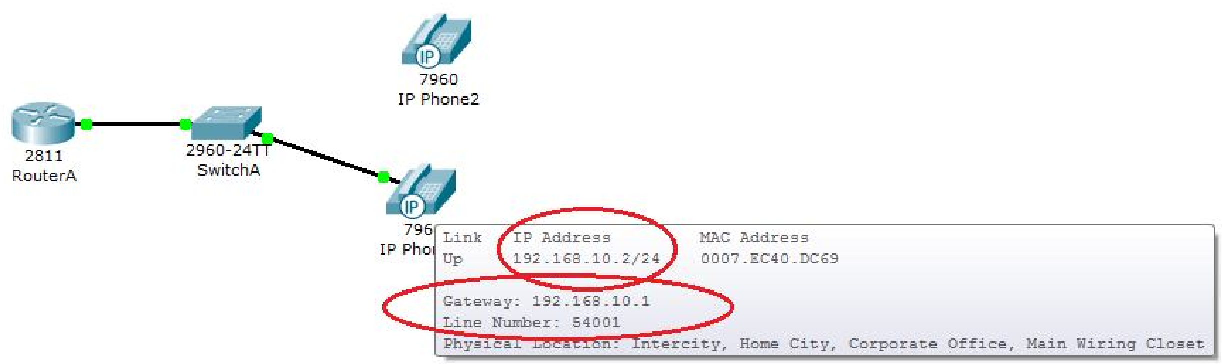


Figure 14.5 IP Address of IP Phone 1 displayed on screen

Figure 14.6 GUI of IP Phone 1 after successful configuration

Task 7: Configure the phone directory for IP Phone 2

Connect IP Phone 2 to SwitchA and power the phone ON using the power adapter (Physical tab).

RouterA(config)#ephone-dn 2 #Defining the first directory entry#

RouterA(config-ephone-dn)#number 54002 #Assign the phone number to this entry#

Task 8: Verify the configuration

Ensure that the IP Phone 2 receives an IP Address and the phone number 54002 from RouterA (this can take a short while). Follow same procedure as task 6.

Dial 54001 and check if IP phone 1 correctly receives the call.

LAB ASSIGNMENT # 01:

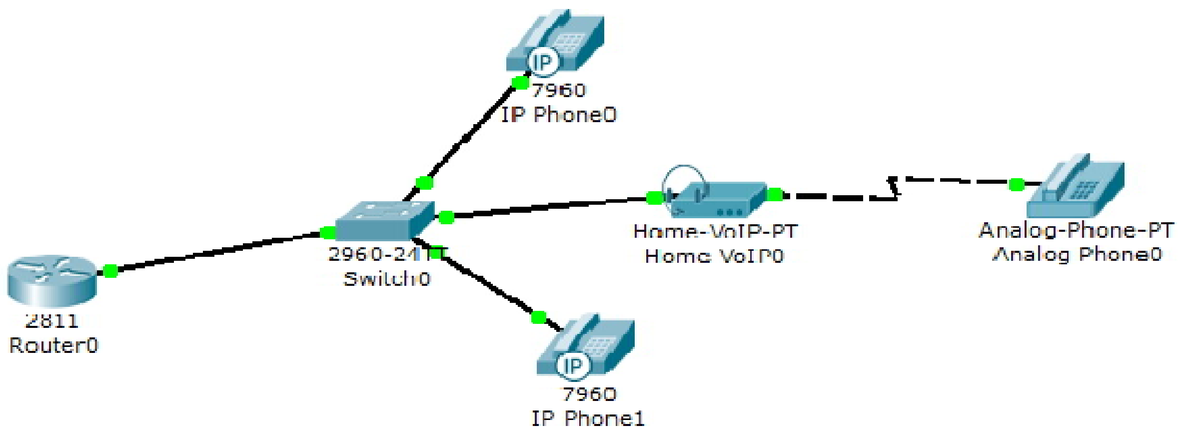
Implement and configure the following topology incorporating analog phone with IP network.

Specifications:

VLAN number: 1

Network Address: 192.168.10.0 255.255.255.0   Port: 2000

Directory numbers: 11111, 22222, 33333



Topology of the network

LAB ASSIGNMENT # 02:

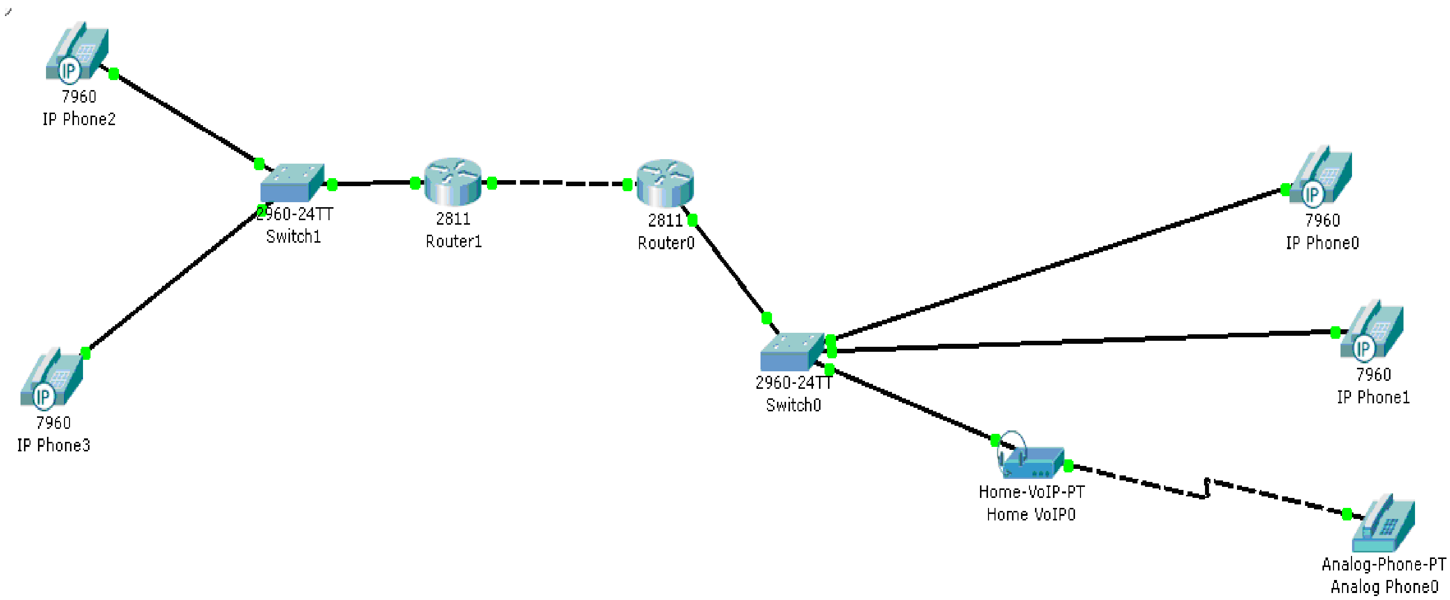
Implement and configure the following topology incorporating analog phone with IP networks.

Specifications:

VLAN number: 1

Network Addressing: as required

Directory numbers: as required



Topology of the Network