Chapter One

Introduction

1.1 Preface

Communication is the backbone of any air traffic control (ATC) activity and the Civil Aviation Authorities (CAAs) are responsible for delivering reliable communication services to airlines for supporting their mission critical applications.

The ideal solution for networks used in air traffic control requiring multiplexing of legacy voice and data systems as well as routing of IP traffic over popular WAN infrastructures or VSAT.

The Multi-service Integrated Access Devices integrates traffic over a variety of enterprise network infrastructures. It can be used over Switched/Lease Lines, Frame Relay and ATM backbones, satellite networks as well as IP backbone. Line costs are reduced by bundling various boundaries of traffic onto a single network infrastructure. [22]

1.2 Problems definition

The existing Air Traffic Control Networks suffer from many problems such as:

- Higher operation cost (Network and support).
- Bigger delay in most sensitive applications (ex.: VHF radio, Hotlines, telephones and radars data).
- Higher latency.
- Higher outage and instability.
- Not all types of legacy traffic can be routed.
- Lower Network availability and most of them doesn't have backup.
- Existing network special in Africa still using the copper as a media.

1.3 Objectives

Implement and evaluate a network Models solution for packetizes the many different interfaces of the ATC applications into variable bit rate data streams to fulfil the Air Traffic Applications Requirements and:

- To reduct in overall operating costs – Network and Support.
- To minimizes latency.
- To Save Bandwidth by Dynamic bandwidth utilization.
- To provide high Network stability and reliability.
- To provide flexible network connections.

- To reducing transmission delays and preserving the quality of delay-sensitive traffic (voice/fax).
- To speeds critical data to its destination.
- To allows any type of legacy traffic to be routed.
- To reducing outage by implement two Networks main and backup.

1.4 Methodology

Practical implementation and evaluation two different networks solutions for ATC environment Specifying parameters applicable to different networks by two systems models

The most important way to exercise and impalement ATC network is to use multiplexing devices supports the ATC applications.

Multiplexer is used to connect the ATCs voice communication technologies such as a radio and the radars data as can be seen in figures below. The infrastructure of such networks is usually a proprietary circuit switched network. This implementation will be based on a packet-switched infrastructure using Internet Protocol (IP) and leased lines because of its advantages regarding flexibility.

There are several advantages of using the open IP standard instead of proprietary circuit-switched solutions. One major advantage is increased interoperability.

1.5 Thesis Outline

The organisation of this thesis is as follows. Starting with a brief description of ATC communication networks, problems definition of existing networks and the methodology in chapter one.

Chapter two mentioned the Background of the old and latest ATC networks technology comparison showing the weakenss and the improvements used in this field are reviewed in detail also in this chapter the main device used in the models is studied and reviewed its features and capability.

Chapter three presents an overview on the two proposed network models of the ATC, are studied and analyzed, and also the ATC applications services protocols are mentioned.

Then the models configuration, implementation and results are shown in chapter four.

And this is followed by conclusion of the network models configuration and implemention and the future work in chapter five.

Chapter Two

Threortical Background

This chapter presents an overview on the ATC networks background comparison between old ATC networks and modern networks based on IP weakness and improvement, also this chapter discussed and studied the main device used to implement the two proposed network models. The key feature and the technology of the multiplexer’s capability, Multiplexers link technology WAN Leased Line, WAN Frame Relay and its Characteristics will be reviewed. Efficient multiplexing and high connectivity techniques In E1 and IP VSAT connectivity studied and analyzed.

2.1 Introduction

Pilot/ATC communication is normally done using voice radios. Initially, the capacity and capabilities offered by this system were sufficient. As air traffic and routes increased and small, private aircraft became more available, this system began to show its weaknesses. The most important ones are air control networks stability and reliability.

Channel congestion is caused by the increasing number of aircraft using the same radio and data source while in the same ATC sector.

As this number increased, the amount of information became more and more difficult to transfer using that old circuit switched networks.[22]

2.1.1 Delays

The main weakness of any packaged voice stream, in comparison with classical digital PCM voice signal, is delay. Once a packet takes longer than 100 - 200 milliseconds to travel to its location, voice quality starts to become unacceptable.

2.1.2 Packet loss

Packet loss can be caused by congestion, router changes as result of faulty network line and sometimes by a too long delay in the network. Lost or dropped packets can be a significant problem affecting voice quality.

2.1.3 Voice compression

A user's voice signal must be sampled and converted to a digital bit stream before it can be transmitted over the network. The use of more compression generally results in worse voice quality, and adds more delay due to the additional processing required.

2.2 Overview of Circuit Switched Networks

Since the air traffic networks are the means with which to guide the aircraft, it is a network of critical importance. Due to the safety critical environment of an airfield, the quality requirements for these networks are strict. The network must provide a reliable service of sufficient quality, there must be no long delays in communication, and sound quality should be adequate. The figure 2-1 shows the current ATC switched Network which is depending on PSTN backbone. [2]

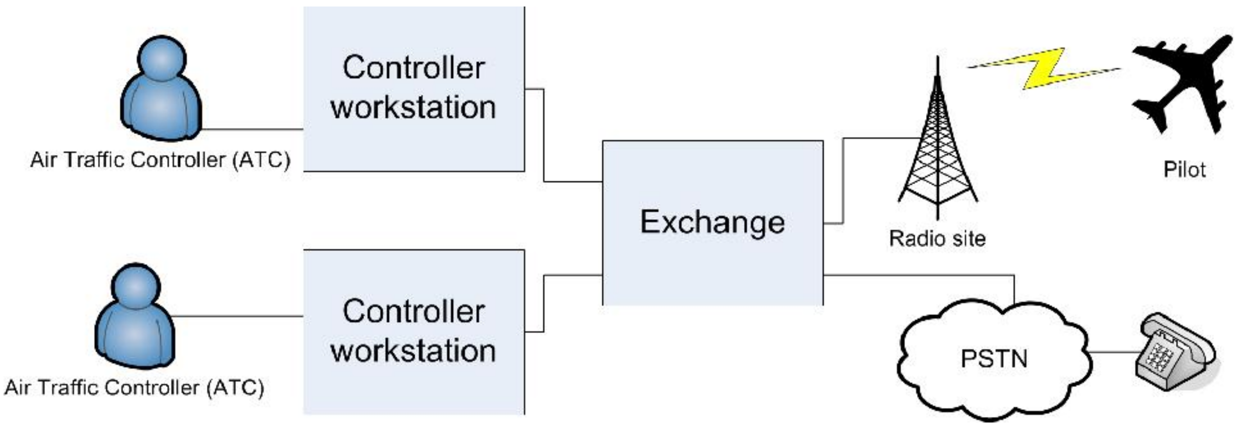


Figure 2.1: General block diagram of old circuit switched ATC network

The controllers can communicate through both radio and telephone using a single work station, called a controller working position (CWP). Using their work stations, the controllers can choose which radio frequencies to receive from and send to, and also make calls both internally within the airport and externally over the PSTN. (See figure 2.1)

2.3 Communication technology—circuits or packets

In recent years, the communications industry has been moving from circuit-switched to packet-switched technology based on the IP. This applies also to voice communications within ATC, where the interest in infrastructures for ATC networks has been shifting from circuit-switching to IP technology.

The modern ATC networks are combined of E1 and IP based VSAT network connected in a full mesh through smart multiplexers, theVSAT network configured as a fully redundant network to the fibber optic network (E1), provides backbone connectivity for VHF voice communications relay to cover the national airspace. The airspace is controlled by their fully redundant ATC master control VSAT station. Furthermore, the VSAT provides geographic redundancy in transporting master control station traffic. In the event of a failure of the master control station the VSAT solution system automatically provides connectivity to one of the remote ATC sites. All sites will support a complex mix of voice and data communications including VHF Extended Range, Hot Lines, Radar, AMHS and ADS-B, metrological data, Intranet, Internet and general corporate internal communications.

Migrating to an IP network allows the support of the older ATC equipment while providing a native IP backbone for new applications and equipment. Combined with VSAT bandwidth savings features of Bandwidth on Demand, Compression, and its latest optional feature - Mesh Adaptive Coding and Modulation (Mesh-ACM Trademark) - provides some of the industry's lowest recurring costs. E1 and IP Backbone becoming the technology of choice for ATC applications as proven by the number of project win in this critical market segment. [10]

2.4 Proposed Networks Models Architecture

These networks tend to require faster, more flexible WAN links and voice and connections, proposed E1 and IP over VSAT solutions are based on higher speed terrestrial link and VoIP connecting modern multiplexing technology.

- Radio and data interface connected by the latest multiplexing devices.
- Multiplexers connected together through E1 leased lines over Fibre infrastructures.
- Or through mesh IP VSAT Network linking meshed multiplexers networks.
- Multiplexers will provide telephony services and IP data services.

2.5 Multiplexers Description

The Multiplexer family of Multi-service Integrated Access Devices integrates traffic over a variety of enterprise network infrastructures. The Multiplexer is both flexible and scalable:

- Interface cards provide more Serial, Analog and Digital connections
- Firmware options to enhance the base feature set
- Interoperates with other network nodes in a variety of packetized environments.

Additionally Multiplexer can be used to create point-to-multipoint networks, either full-satellite or hybrid terrestrial-satellite, transporting voice and data over satellite, perfect for linking remote sites in which terrestrial services are not readily available.

To further increase the performance over satellite, the TCP Acceleration option is also available to overcome the throughput limitation inherent with high latency links.

The Multiplexer does not only support voice over Packet Networks, but also includes excellent LAN, SNA and legacy data support. [17]

2.5.1 Hardware Platforms

•The SDM-9230 standalone chassis model for branch, regional and remote offices The SDM-8400 standalone chassis model, a data-only integrated access device that supports multiple serial ports and Ethernet for increased serial port capacity. These Multiplexer hardware platforms employ the following standard and proprietary technologies:

•Standard: IP/Ethernet circuits, IP RIP v1/v2, VoIP using SIP

•Proprietary: full support of standard voice algorithms and both standard and proprietary data protocols, PVCR connections to converge voice and data, Voice over PVCR, Frame Relay and IP -based satellite connections. [17]

2.5.2 Multiplexers Chassis Comparison

Table 2-1: Multiplexers Chassis comparison

	SDM-8400	SDM-9230
T1/E1 Ports	-	6
BRI S/T physical ports	-	6
Digital Voice Ports	-	120
Analog Voice Ports	-	12
FXS	-	12
E&M	-	12

2.5.3 SDM-8400 Description

The Multiplexer SDM-8400 showing in figure 2-2 enables incremental scalability of existing multiplexer multifunctional routers by adding serial port capacity. It can also be used as a standalone

Unit for data only applications.

Figure 2-2: Multiplexer SDM-8400 unit

Interfaces

- One 10/100Base-T Ethernet (RJ45 connector)
- Four or eight serial ports (HD26 female connectors)

2.5.4 SDM-9230 Description

The Multiplexer SDM-9230 is a standalone chassis designed for flexible convergence at the branch office level figure 2-3.

It is a scalable product for branch offices that require support for 12 analog or 120 digital telephony channels and the ATM option.

Figure 2-3: Front view of the SDM-9230 unit

Interfaces

- One DSP connector per unit (DSP-160 type or High Density DSP module)
- Three expansion slots for the interface cards
- Analog voice interface cards: 2 and 4 ports FXS, 2 and 4 ports E&M
- Digital voice/data interface cards: single or dual port T1/E1 120 ohms,
- One Serial port interface.
- Two Ethernet port interface.

2.6 WAN Leased Lines

2.6.1 About PVCR Technology

PVCR technology has been designed to carry multiple traffic types (data, voice, fax and LAN) over a Wide Area Network. The advantages of PVCR include:

- Efficient multiplexing of data from all sources
- High-performance data compression to improve throughput and reduce telecommunications costs
- Load balancing to provide high-speed support using multiple circuits
- Dynamic bandwidth utilization and flexible network connections
- Enhanced rerouting capabilities, with switching through intermediate nodes
- Dial backup of WAN links
- Bandwidth On Demand over leased lines for cost-effective management of bursty LAN traffic
- Efficient prioritization across the network, reducing transmission delays and preserving the quality of delay-sensitive traffic (voice/fax).

2.6.2 PVCR Principle of Operation

PVCR operation includes several distinct functions as mentioned in figure 2-4:

- **Protocol Sorting:** First, whole frames are differentiated as to traffic type, and identified as to priority level
- **Fragmentation:** The Multiplexer splits the frames into small cells, or blocks, to reduce access
- **Data Compression:** After fragmentation it compresses the contents of the cells, when required for smooth traffic flow: The Multiplexer then multiplexes cells from all sources
- **Transmission: Finally**, it transmits the multiplexed, compressed traffic across the link The Multiplexer at the receiving end performs the reverse process: demultiplexing and decompressing the cells, then reconstructing the original whole frames before sending them to the appropriate destination user.

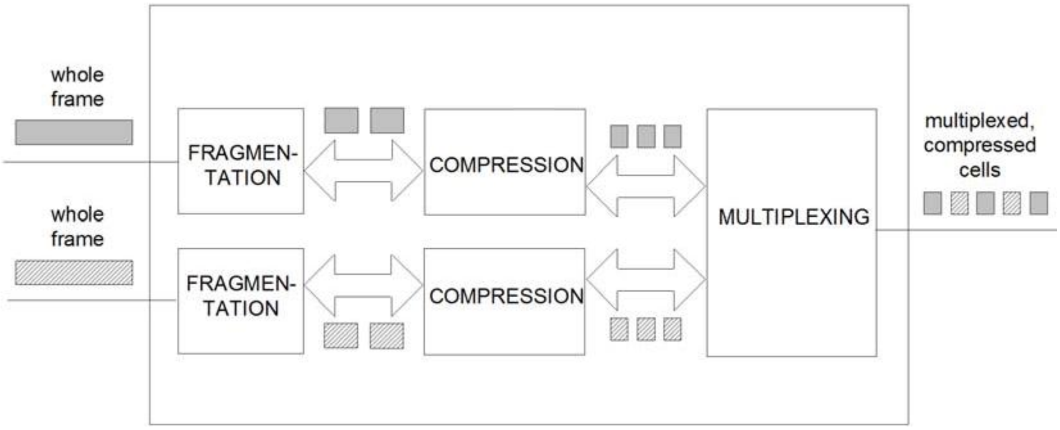


Figure 2-4: PVCR Principle of Operation

This process incurs very little network delay. The Multiplexer can switch up to 8000 cps

2.6.3 Protocol Sorting

Through protocol sorting, Multiplexer differentiates delay-sensitive traffic from non-sensitive traffic, and directs these traffic streams to their corresponding transmit queues. Sorting is based on the traffic type, the weight class to which the traffic is assigned, and any additional traffic filters you may wish to create.

The architecture of the Multiplexer receivers combined with their capacity to differentiate traffic types, ensures accelerated processing of delay-sensitive data (for example, voice) without jeopardizing the processing of non-sensitive data.

2.6.4 Fragmentation

During fragmentation, the Multiplexer splits whole frames into small cells to reduce access time. It also performs protocol stripping at this stage, to remove flags and synchronization characters from transparent user data and dynamically compresses the cell contents in accordance with current traffic levels.

Frames received on all ports are analyzed and cut into variable-length cells, from 1 to 96 characters long. A small frame with 96 characters or less occupies a single cell. Larger frames are segmented and the resulting cells tagged with sequencing information for later reconstruction.

During fragmentation, the order in which the Multiplexer cuts the frames depends on:

- The order the frames arrive in
- The priority assigned to that traffic type
- The priority of other traffic types arriving at the same time.

If a high-priority frame (for example, voice or SNA) arrives while the Multiplexer is fragmenting a low-priority frame (for example, LAN), it turns immediately to the high-priority frame before continuing.

From that point on, cells are cut alternately from the two frames (high and low-priority) as the traffic arrives at the receiver.

There are six types of traffic cells:

- BEGIN: First cell of a long frame (more than 96 characters)
- MIDDLE: Middle cell of a long frame
- END: Last cell of a good frame
- ABORT: Last cell of a bad frame
- COMPLETE: The only cell of a small frame (96 characters or less)
- VARIABLE: A compressed, variable-length

For example, a long frame will be fragmented into the following cells if it is a good frame: see figure 2-5

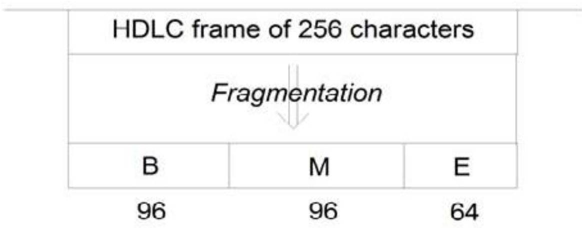


Figure 2-5: A long frame fragmented into cells

Through fragmentation, the Multiplexer is able to reduce internal delays in three ways:

• Compression and multiplexing can start as soon as the first cell is received. The Multiplexer does not have to wait for the end of frame. This reduces transmission delay to the time needed to receive and cut a single cell.

• Fallback transmits and receives clocks. If the port or channel is in fallback mode, the Multiplexer reduces the transmitter speed to an appropriate fallback speed when the number of characters left in an incompletely fragmented frame falls below a predefined threshold.

• Early transmission. The Multiplexer starts transmitting frames to the connected device as soon the first few cells are delivered to the transmitter, even if the frame has not been fully fragmented yet.

2.6.5 Data Compression

If traffic levels are high Multiplexer applies a dynamic data compression algorithm to the cell contents for increased throughput. After compression, each cell contains the amount of data required to maintain a smooth, delay-free flow. Cells do not have to be full before they are transmitted.

Data compression improves bandwidth utilization, which results in reduced telecommunications costs. It is applied to all protocols that analyze the traffic in terms of frames.

Voice compression algorithms operate independently of data compression, and use DSPs for processing.

The built-in data compressor on the Multiplexer can enhance the throughput of a composite link from 100% to 400%. The actual compression ratio achieved depends on the type of data being transferred and the amount of traffic on the link. It is usually between 2 and 4.

The Multiplexer compressor processes the cells that were produced during fragmentation, in the following way:

• First, it carries out run-length encoding on the contents of each cell, replacing identical characters with an escape code.

• Then it builds a dynamic dictionary for each port, using the most frequently encountered character strings.

- It then assigns a Huffman code to each common string, and
- Replaces the character strings with these codes.

Decompression is carried out by the data compressor in the destination multiplexer unit. It replaces the Huffman codes with the original character strings, and decodes the escape codes as the original characters.

2.6.6 Multiplexing and Cell Relay Transmission

Using a process called Programmable Variable Cell Relay (PVCR); the Multiplexer sends the individual, variable-sized cells resulting from fragmentation and compression over the high-speed link. Delay-sensitive cells such as voice and fax go first. User-defined criteria determine traffic prioritization for LAN and legacy data.

The mix of high and low-priority cells ensures that, before a non-sensitive frame is sent, a delay-sensitive frame has been completely transferred to the remote device. This avoids interruptions in voice/fax traffic and session timeouts for data traffic such as SNA/SDLC.

The Multiplexer uses an advanced statistical multiplexing technique that combines voice, fax and data from all active input ports and channels, and sends the combined traffic over the network via one or more composite links.

• When multiple links are available, load balancing ensures that an equal traffic weight is transmitted over all available links.

• A multiple link configuration also allows for automatic link backup, which provides additional reliability in case of link failure.

• Through Bandwidth On Demand, multiple links can be activated and deactivated automatically according to current traffic levels and bandwidth needs. This is ideal for applications requiring varying bandwidth levels

• You can also define a range of different link connection scenarios for automatic time-of-day connects; building a schedule that mimics your peak and minimum traffic times.

Multiplexer WAN links use a proprietary level 2 synchronous protocol called PVCR (Programmable Variable Cell Relay) and include proprietary level 3 protocols.

• Level 2 layer: Responsible for retransmitting frames in case of errors on the links (e.g. for SNA devices). Also used when reliable asynchronous transmission is required.

When the link detects errors on cells destined for a transparent user port, the Multiplexer resets all queues for that port, resynchronizes the compressor and then resumes transmission.

• Level 3 layer: Includes flow control and traffic congestion procedures. Also responsible for establishing and maintaining up to 64 channels in an SNA/SDLC application, one per SNA device (PU).

The composite link carries HDLC frames, with multiple cells in each frame. The frame size on the link is limited to 1024 bytes.

This has no bearing on the frame size permitted on transparent user ports or channels configured with the HDLC protocol.

The Multiplexer at the remote end demultiplexes the cells and uses the cell numbering to sort the cells into their original order. It reconstructs the original frames from this cell sequence, and transmits them to the destination devices.

2.6.7 Mesh Topology

Multiplexer nodes can be connected in a WAN mesh topology. In the application represented in Figure 2-6, source unit A and destination unit D perform protocol sorting, fragmentation, compression/decompression, multiplexing/demultiplexing and cell relay transmission. The intermediary units B and C perform cell relay transmission only.

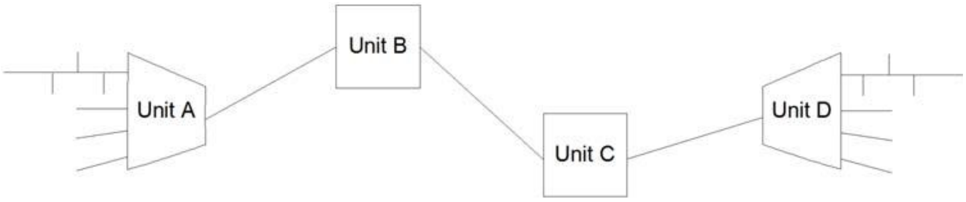


Figure 2-6: Mesh Topology

2.6.8 Prioritization of Voice/Fax Transmissions

The Multiplexer is well suited for handling voice and fax traffic, due to its ability to prioritize traffic, reduce delays and control traffic congestion. When data transmission is cell-based, the frame does not have to be completely processed at any location before changing to a new virtual path. The result is that latency and overhead are reduced to a minimum, producing shorter transmission delays. This is essential to satisfactory voice and fax communications.

Due to the delay-sensitive nature of voice and fax transmissions, it is essential that they be prioritized to avoid the generation of delays. All voice/fax traffic entering a multiplexer channel is automatically given high-priority status with respect to data from other sources. This ensures guaranteed bandwidth access in a shared network environment, and an uninterrupted flow from source to destination.

Voice/fax traffic can be assigned to a specific priority class if desired.

Each Multiplexer along the virtual connection path selects a virtual channel according to the priority class assigned to that virtual connection. In this way, the high priority of voice/ fax traffic can be maintained uniformly from source to destination.

Each Multiplexer unit monitors it’s transmit queue for the priority level of the cells that are received. When high-priority cells arrive, they are expedited to the next Multiplexer on the virtual path before processing the lower priority cells. Lower-priority cells are buffered until the higher-priority voice and fax cells are sent. This also guarantees voice/ fax prioritization throughout the network.

2.6.9 Intelligent Congestion Control

The Multiplexer is equipped with intelligent congestion control that prevents adverse network situations from interfering with delay-sensitive traffic such as voice communications.

Traffic can be sent from the source multiplexer unit over several hops, or intervening Multiplexers, to reach the destination unit. The best route is always chosen, and rerouting is performed automatically through simple routing table updates. As a result, the network topology is highly flexible and adaptable to congestion.

Delays tend to get higher as a virtual path gets longer, each hop typically adding a 10 ms delay. If any Multiplexer along the virtual connection detects congestion while it is transmitting, it automatically retains more data in its buffers to ease the flow.

The turnaround time for the Multiplexer is typically 150 ms across a Wide Area Network. This is significantly better than satellite service, which a typical latency of 250 ms. has given that the human tolerance for voice delay is about 500 ms; the processing speed of the Multiplexer produces satisfactory results even when congestion is encountered.

The Multiplexer includes flow control techniques to minimize congestion problems and ensure maximum efficiency of the transmitter and receiver at all times. These techniques include:

- Fallback speed of the transmit and receive clocks, and
- Early frame transmission based on the Transmission start level.

2.6.10 Fallback Speed

The Multiplexer supports fallback for data received over a transparent user port (non-PVCR) under the following conditions:

• The transparent user port is a built-in serial port Digital channel and ports on the Dual Serial interface card do not support fall- back.

• The Multiplexer controls both the transmit and receive clocks on the port (DCE port in internal clocking mode)

With DCE internal clocking, the Multiplexer can set the port speed (normal or fallback) depending on current traffic conditions (uncongested or congested, respectively).

• The equipment connected to the port supports fallback speeds.

Fallback is enabled or disabled with the Fallback speed port parameter. However, the actual fallback speed is not configurable. The Multiplexer determines the appropriate fallback speed automatically from the combined speed of all WAN links that carry traffic to the destination unit.

The Multiplexer provides two independent clocks: one for the receiver (the receive clock) and one for the transmitter (the transmit clock).

When fallback is applied on the receive clock, the speed of the transmit clock is not affected. Likewise, when fallback is applied on the transmit clock, the receive clock is not affected.

Fallback on Receive Clock

Congestion can occur on the WAN links when the Multiplexer transmits data to the remote side. This congestion is caused by a low compression ratio or too much traffic queued by the user ports. When congestion occurs, a burst of data on a user port will be stored in the multiplexer receiver queue.

However, if the data burst size exceeds the capacity of the receiver queue, that data will be lost. This situation is called a receiver overrun. When a receiver overrun occurs, the Multiplexer requests the attached equipment to retransmit the data.

Overruns occur when the user port speed is too high for current capabilities of all WAN links to the destination unit. To prevent overruns, the Multiplexer slows down the receive clock on the user port to the fallback speed, allowing the WAN links to catch up to the traffic flow.

As mentioned earlier, the fallback speed is adjusted automatically to the current traffic situation. In most situations, the Multiplexer sets the fallback to one half of the sum of the speeds of all WAN links to the same destination unit as represented in figure 2-7 and table 2-2

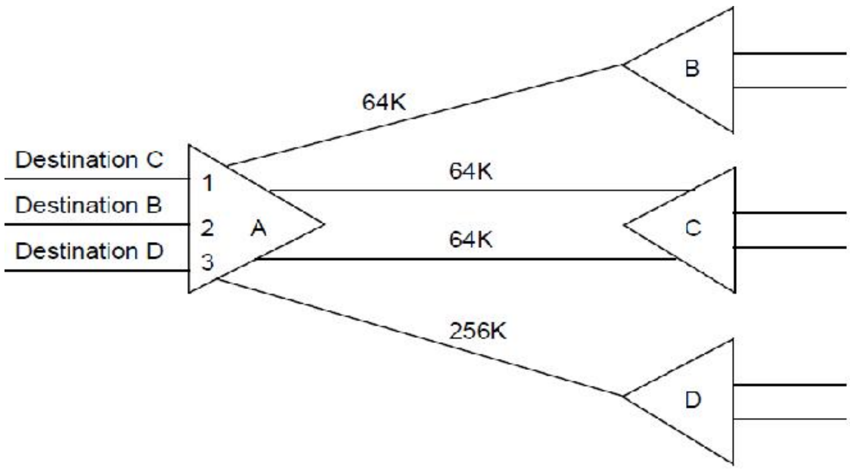


Figure 2-7: Using Fallback Speed for Flow Control

In the example above, the fallback speeds on ports 1 to 3 are calculated as follows:

Table 2-2: Fallback speeds on ports 1 to 3

Port Number on Unit "A"	Destination Unit	Link(s) to Destination	Sum of Link Speeds	Fallback Speed (1/2 the Sum of Link Speeds)
1	C	A - C (dual link)	128K	64K
2	B	A - B (single link)	64K	32K
3	D	A - D (single link)	256K	128K

How it works: When the Fallback speed port parameter is enabled, the receiver reduces its speed if the number of characters in the receiver queue exceeds a configurable threshold (the Transmission start level port parameter). During this time, the amount of data transmitted on the links exceeds the amount of data received on the port, and the congestion situation is resolved before the user port receiver queue becomes full. This avoids both data loss and the delays due to retransmissions. The user port returns to its normal speed when the number of characters in the receiver queue falls below the *Transmission start level*.

Fallback on Transmit Clock

Fallback is applied to the transmit clock to prevent transmitter underruns. Underruns occur when frames arrive too slowly from the remote unit, so that the user port transmits the beginning part of the frame before it has received the end of the frame. When an underrun occurs on the transmitter, the current frame is aborted.

To determine whether underruns have occurred on a particular port, execute the Display Errors (DE) command.

To prevent underruns, the transmitter is slowed down to the fallback speed, which reduces the rate that data is received from the WAN link.

How it works: When the Fallback speed port parameter is enabled, the user port transmitter reduces its speed if the number of characters of an incomplete frame in the output queue goes below the current value of the Transmission start level parameter (see next section). During this time, the port transmission rate is slower than the rate at which data is received from the WAN links, which allows the port to receive the end of frame before the beginning part of the frame has been completed transmitted to the attached equipment. Once the frame is sent, the transmitter returns to normal speed.

2.6.11 Transmission Start Level

A Multiplexer user port can start transmitting a frame as soon as a specified number of characters have been received from the WAN link. Transmission delays are reduced when the transmitter can start before the incoming frame has been completely received.

Transmission flow control must also be applied to avoid underruns and ensure a smooth traffic flow. To specify the number of characters that must be received before the port can start transmitting a frame, set the Transmission start level port parameter.

The user port must be in DCE internal clocking mode, and the attached equipment must support this feature.

In the port transmitter queue, transmission will start when the Transmission start level is reached or the frame is complete. The transmitter will go into fallback if the frame is incomplete and the receive rate is slow

Setting the Transmission Start Level

Set the Transmission start level to AUTO if you would like the Multiplexer to start the user port transmitter automatically according to the port clocking mode, speed and fallback speed settings.

If you do not want the Multiplexer to control the Transmission start level automatically, you need to consider:

- The user port clocking mode

Set Transmission start level to MAX if the port is not in DCE internal clocking mode. The transmitter will start only when the complete frame has been received from the remote unit.

- Whether Fallback speed is enabled on the port

- Set Transmission start level to MAX if fallback is disabled.

If fallback is enabled and the user port is in DCE internal clocking mode, set the Transmission start level according to the speed of the port, as shown in Table 2-3.

Table 2-3: Settings for Transmission Start Level

User Port Speed	Transmission Start Level
less than 56 Kbps	48 bytes
56 - 96 Kbps	96 bytes
96 - 144 Kbps	144 bytes
144 - 192 Kbps	192 bytes
192 - 256 Kbps	256 bytes
256 - 512 Kbps	512 bytes
512 - 1024 Kbps	1024 bytes
1024 - 2048 Kbps	2048 bytes
more than 2048 Kbps	MAX

2.6.12 Multiplexer Multiple Link Capabilities

2.6.12.1 About Multiple Link Capabilities

The Multiplexer supports multiple WAN links with:

- Load balancing (see next section)
- Inverse multiplexing
- Automatic dial backup
- Bandwidth On Demand
- Schedule operation

2.6.13 Load Balancing

The Multiplexer uses a cell balancing method as described in figure 2-8 to accomplish load balancing dynamically across multiple WAN links:

- As described the Multiplexer cuts all frames into small cells, and routes them over the first available link.
- Using dynamic load balancing, the Multiplexer transmits the cells in parallel over all active links.
- Since each cell is routed individually the available bandwidth is maximized and transmission time is reduced as much as possible, resulting in improved response time and throughput.

• The cells are later reassembled in their original frame format by the Multiplexer at the other end.

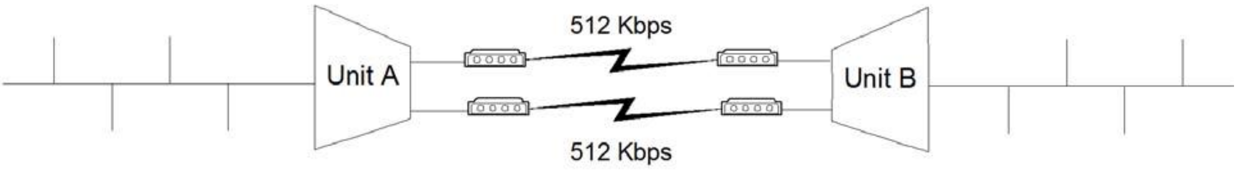


Figure 2-8: Load Balancing Application

In some cases static load balancing (rather than dynamic) should be used in order to set a preferred route over multiple dedicated links. This may be required when:

- More than one route is available
- The cost is greater than 0
- All available routes are equal in cost

You can set a separate preferred route for each traffic class. When you assign a different preferred route to different classes, you ensure static load balancing of transparent user traffic over all routes.

2.6.14 Inverse Multiplexing

The multiple link capabilities of the Multiplexer can be used to create an inverse multiplexing application as in showing in figure 2-9. Inverse multiplexing provides high-speed support using multiple digital circuits. Here are two possible inverse multiplexing applications that can be created on a single Multiplexer:

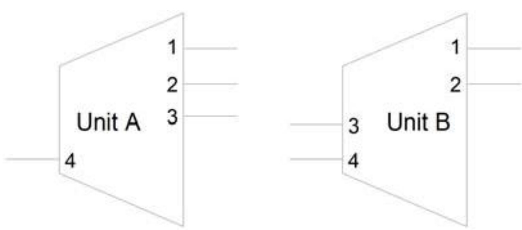


Figure 2-9: Inverse Multiplexing Applications

2.6.15 Bandwidth On Demand

Bandwidth On Demand (BOD) allows for automatic activation and deactivation of multiple WAN links according to current traffic needs. With Bandwidth On Demand, the Multiplexer increases the bandwidth when traffic increases, and decreases it when traffic returns to normal levels.

A BOD solution is ideal for applications that require varying bandwidth levels, for example, LANs with traffic bursts. You can design the network to handle the average traffic load, and allow extra bandwidth to be added only when it is needed.

Bandwidth On Demand is available on dedicated WAN links only, that is, a serial port or digital channel configured with in the PVCR protocol.

How it works: In a typical application of Bandwidth On Demand:

• One WAN link is configured for operation over a dedicated line, for example, local port 1 set in PVCR protocol with DEDICATED mode, connected to remote port 1 set in PVCR protocol with DEDICATED mode.

• A second WAN link is configured for operation over switched circuits, for example, port 2 set in PVCR protocol with BOD-CALL mode, and remote port 2 set in PVCR protocol with ANSWER mode.

• This link is activated only when the bandwidth usage reaches a predefined level (Configured with the BOD level parameter).

• When the bandwidth usage on the dedicated link (port 1) reaches the BOD level and stays at that level for a pre-defined period (configured with the Delay before BOD call activation parameter), the switched line is automatically activated (port2).

The Delay before BOD call activation allows time for the bandwidth usage to return to normal in case of a brief burst of data.

• The traffic load is shared between the BOD link and all other active WAN links.

In other words, load balancing continues to operate between all active links, whether they are dedicated or BOD links.

• When the bandwidth usage decreases to a significantly lower level (about half of the BOD level) and stays at that level for a predefined period (configured with the Delay before BOD call deactivation parameter), the switched line is disconnected. [17]

2.7 WAN Frame Relay

2.7.1 Packet Mode Interface

Frame Relay is a packet mode interface specification that provides a signalling and data transfer mechanism between data equipment and a network. It allows LANs located far apart from each other to be interconnected with a high-speed WAN protocol.

Each frame (or packet) contains header information that influences the routing of the data to the desired destination. The Multiplexer can concentrate Frame Relay traffic originating from multiple devices, local or remote, onto a single Frame Relay connection. The Multiplexer also supports Frame Relay over IP. This permits using the Multiplexer PVCR protocol to integrate voice and data over the Internet. With FRoIP, the Multiplexer routes a PVC connection over IP instead of Frame Relay

A Frame Relay network uses virtual circuits, which are logical paths established between two network access points. A Permanent Virtual Circuit (PVC) is established using administrative procedures, typically a network manager who configures the network using a network management facility. This type of virtual circuit remains in the network until the network manager removes it. [1]

2.7.2 Permanent Virtual Circuits

The Multiplexer supports the Frame Relay interface using multiple PVCs. These PVCs are linked to different locations, bundled and attached to the same physical connection to the Frame Relay network. Frame Relay packets are routed over the network via the PVCs according to the address information embedded in the frame header.

No error correction is done by the network. The responsibility to retransmit is left to the user equipment.

The address information in the frame header is known as the Data Link Connection Identifier (DLCI), which is provided by the carrier. A separate DLCI address is required for each PVC used by the Multiplexer. The Multiplexer encloses the DLCI address, representing the destination of the frame, in a two-byte field header. The carrier can then route the frame to the proper destination.

The carrier also provides notification of any congestion conditions in the network by setting the Forward and Backward Explicit Congestion Notification bits (FECN and BECN) in the header of the routed frame. The Multiplexer monitors these bits in the header frame to determine when congestion control is required. This is particularly important for Voice over Frame Relay transmissions, where the Multiplexer can perform voice fallback based on the status of the BECN bits.

On each Frame Relay port of the Multiplexer, two DLCI addresses are reserved for the Frame Relay Management Interface Protocol. DLCI address 1023 is used for the LMI (Local Management Interface) protocol, and DLCI address 0 is used for the ANNEX-D protocol, which is part of the ANSI standard, or the Q.933 protocol, which is part of the CCITT standard. These management interface protocols provide a polling mechanism for requesting status information on [20]

CHAPTER THREE

Network Models Description

In this chapter, analytical Network Models the operating theoretically and the application Protocols used, are mentioned and reviewed , most important parameters and their values for setting up multiplexers network models link over E1 link and VSAT Networks, are studied in details E1 interface, E1 signalling, channel assignment, coding scheme and PVCR protocol link management .

3.1 E1 Model Description

3.1.1 Using E1 link technology for connecting Multiplexers

Below is E1 model description using block diagram of three multiplexers connected to each other using E1 technology

Model layout:

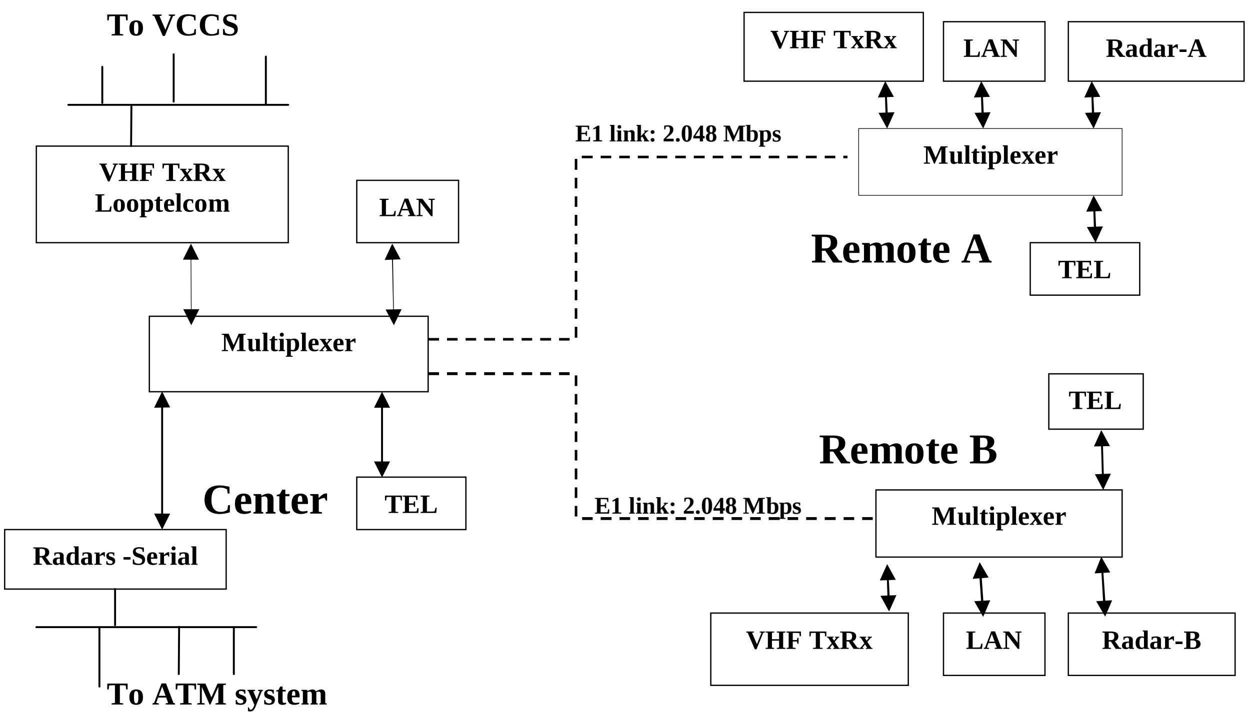


Figure 3-1: Diagram of E1 Model Link Solution

3.1.2 About Digital Connection

The following trends are commonly experienced by enterprise networks:

- A growing interest for integrating voice and data traffic,
- An ever increasing need for greater bandwidth.

As a result, these networks tend to require faster, more flexible WAN links and voice connections. T1 and E1 lines provide the throughput required. In addition to greater speed, they also provide greater flexibility, as they can be subdivided into channels of varying sizes. T1 and E1 lines have become one of the most common interfaces in public voice/ data networks.

The Multiplexer optionally provides T1, E1 and ISDN-BRI channelized digital connections that can be used to connect a central site to multiple remote locations. Each channelized interface module supplies a number of individual channels, each of which can be terminated in a separate digital circuit at a different remote site. No external CSU/DSUs or associated cabling are required at the central site. [17][20]

3.1.3 Multiplexer Digital Interfaces

A channelized digital connection on the Multiplexer can be established using a T1, E1, ISDN-BRI or E1/T1 (Dual Framer) interface. On most Multiplex products these interfaces are provided on optional interface cards which are installed in slots on the base unit.

3.1.4 E1 Interfaces

The Multiplexer supports E1 at 75 and 120 Ohms:

• E1-75 Interface: A 2.048 Mbps interface at 75 Ohms supporting up to 32 times- lots per port, with two coaxial BNC connectors per port

• E1-120 Interface: A 2.048 Mbps interface at 120 Ohms supporting up to 32 timeslots per port, with one RJ-45 connector per port.

3.1.5 E1 Channel Assignment

E1 has 32 channels in its data service, each at 64 Kbps. Of these channels, only 30 are used: channels 1 to 15 and 17 to 31. A channel occupies an integer number of timeslots, and the same timeslot positions in every frame. Each of channels 1 to 15 and 17 to 31 can accommodate, for example, a PCM-encoded voice band signal or a 64 Kbps digital data signal.

The D-channel, when present, is assigned timeslot 16 (which is reserved for this purpose). Contiguous HDLC flags are transmitted on the D-channel when there are no frames to send. [20]

3.1.5.1 E1 Error Detection

The following functions are available for E1 transmissions:

• Multiframe Alignment Signal: In E1 transmissions, a multiframe is made up of

16 consecutive frames numbered from 0 to 15. The multiframe alignment signal is 0000, which occupies bits 1 to 4 of channel 16 in frame 0.

• Signalling Channels: Channels 1 to 15 and 17 to 31 are assigned to telephone channels numbered from 1 to 30.

• Frame Alignment Signal (FAS): Channel 0 carries the FAS, which is used for alignment purposes. It occupies positions 2 to 8 in timeslot 0 of every frame. In order to have synchronization the FAS code, 0011011, alternates with the Non- FAS byte, which contains a x1xxxxxx.

• Cyclic Redundancy Check (CRC): A 4-bit CRC can be created from other bits in timeslot 0.

• PVCR Link Management: The detection and retransmission of RAI and AIS alarm signals over an E1/T1 interface if the PVCR (WAN) link is down.

3.1.5.2 E1 Coding Scheme

The coding scheme of the E1 interface is called “high-density bipolar - 3 zeros”, or HDB3. As with the T1 coding scheme described earlier, HDB3 is based on bipolar encoding. The scheme replaces strings of four zeros with sequences containing one or two pulses. In each case, the fourth zero is replaced with a code violation.

To avoid introducing a DC component, another rule ensures that successive violations are of alternate polarity. Before replacing the zero sequence, the scheme determines whether the number of pulses since the last violation is even or odd, then determines the polarity of the last pulse before the occurrence of the four zeros.

Codes are also specified for idle channels and idle slots. A pattern including at least three binary ones in an octet must be transmitted in every timeslot that is not assigned to a channel. These are timeslots that may be awaiting channel assignment on a per-call basis, residual slots on an interface that is not fully loaded, and so on. The same pattern must appear in every timeslot of a channel that is not allocated to a call in both directions.

On the Multiplexer, the HDB3 scheme can be turned off through a configurable parameter.

These zero suppression coding schemes should not be confused with PCM encoding methods, which are used to digitize analog voice into digital voice. The Multiplexer supports both µ-law (mu-law) for installations in North America (typically T1), and A-law for installations in Europe (typically E1).

3.1.6 PVCR Link Management

The Multiplexer is able to send AIS (Alarm Indication Signal) or RAI (Remote Alarm Indication) alarm signal over an E1/T1 interface if the PVCR (WAN) link goes down. The AIS or RAI alarm indicates that this link cannot be used.

In addition, the Multiplexer can detect an AIS/RAI signal on an interface and transmit it to a remote unit, which is then able to retransmit the AIS or RAI to the user endpoint.

For example, if the remote unit is lost, the Multiplexer will generate an alarm to advise the PBX that it must not continue to transmit on this interface.

This scenario is also supported on an E1 interface between MSCs in a GS application, or in any type of DCME application.

• When a WAN link goes down, the Multiplexer disables the E1 interface on the local side if the peer E1 interface is down on the remote Multiplexer.

• In this way there is a propagation of the failure all the way to the E1 interfaces of the MSC, which will stop routing calls through that set of E1 channels.

Example: Unit A is connected via PVCR links to Units B and C. The digital interface in slot 1 is configured to reach Unit B and the digital interface in slot 2 is configured to reach Unit C. If the link to Unit B goes down, an alarm will be generated only on the second port of the E1 interface card in slot 1. [19]

3.2 IP VSAT Model Description

3.2.1 Using IP over C-Band VSAT to connecting multiplexers

Introduction

Dedicated VSAT ATC communication networks provide virtually error-free, carrier-grade (99.9% network reliability) digital voice and data communications services.

Ideal solution for networks requiring multiplexing of legacy voice and data systems as well as routing of IP traffic over VSAT

Fibber optic outages are quite frequent in certain parts of the world

Multiplexer allows for cost-effective backup while preserving traffic integrity and reducing bandwidth

Compress VHF voice channels with LDCD

Compression of traffic over WAN

In Air Traffic Control networks transmission of audio should be synchronized: With LDC the Multiplexers will automatically compensate for the largest delay introduced by the satellite back up link.

Network jitter will be automatically compensated for by continuous measurement of network delay. [15] [24]

Below is IP VSAT model description using block diagram of three multiplexers connected to each other using full meshed three VSAT modems network

Model layout:

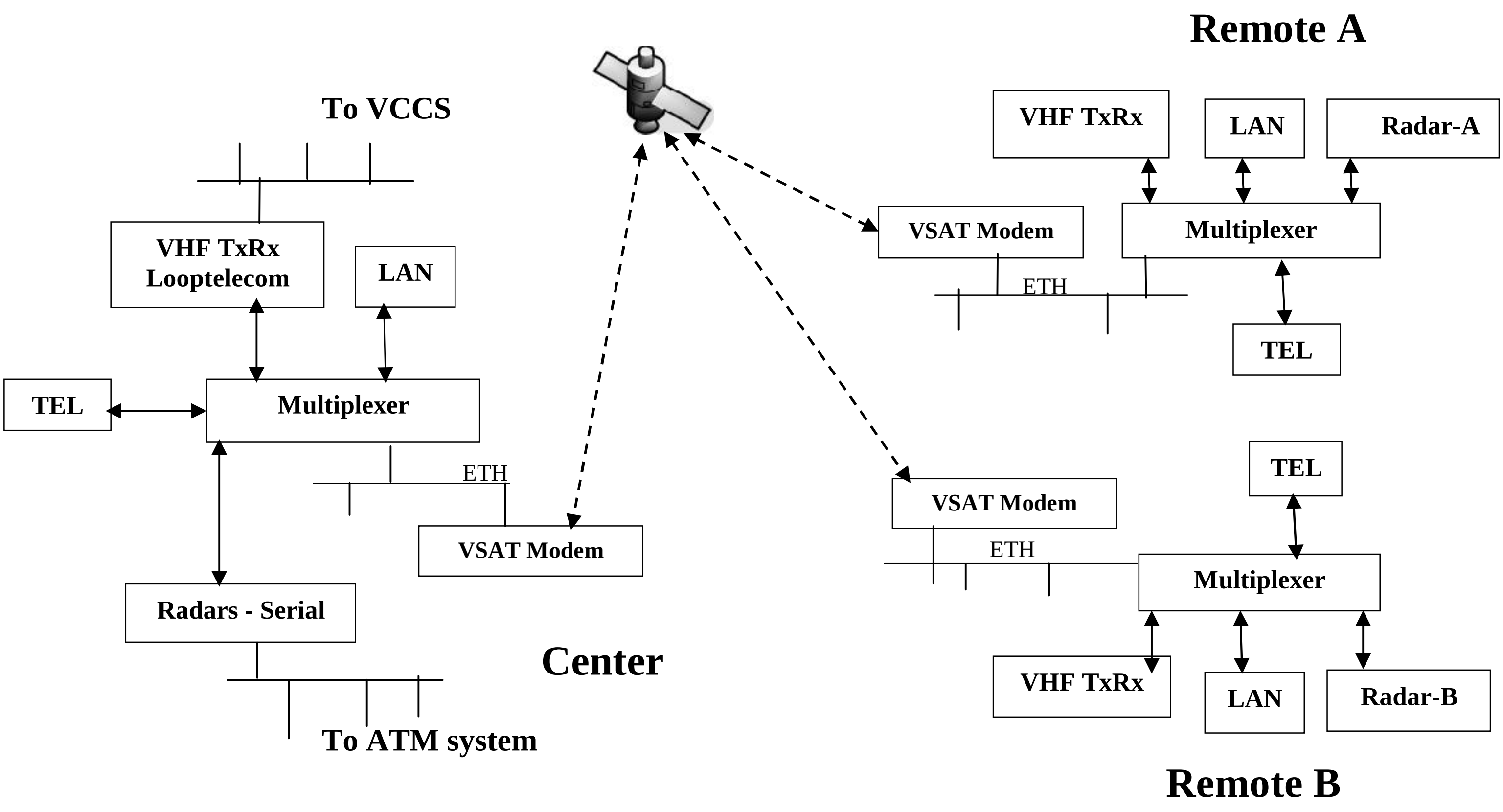


Figure 3-2: Diagram of Model Two full meshed VSAT Solution

3.2.1.1 About Multiplexer Link over VSAT

• With VSAT Modem, a hybrid terrestrial/satellite network is created using a single hardware platform and network management system.

• This solution is able to exploit the broadcast nature of satellite communications, while taking advantage of powerful the Multiplexer features.

The Multiplexer network over IP VSAT solution requires external third-party satellite modems that operate up to 2 or 6 Mbps, depending on the application.

3.2.1.2 Multiplexer Link over VSAT Features

The Multiplexer connected through Satellite modem based on IP standards and offers a scalable solution for both remote and central sites:

• Efficiently consolidates voice, data and LAN traffic with terrestrial networks and equipment.

• Offers a hub less VSAT solution that requires neither an expensive DAMA

Neither computer nor a central site switches (TDMA).

• Interfaces with all third-party satellite modems.

• Supports a wide variety of satellite network topologies: single or distributed star, partially meshed, point-to-point and multipoint networks.

3.2.1.3 Multiplexer Platform

To support satellite functionality in a multiplexer network, the Special software is loaded onto each participating multiplexer product.

• Supports IP/IPX/OSPF routing and bridging.

• Takes advantage of highly efficient data compression algorithms.

• Adds provision for line failure with Virtual Connections and Dial Backup functions.

• Optimizes bandwidth utilization with cell-based multi-protocol prioritization, Bandwidth-On-Demand and Load Balancing.

• Manages the impact of bursty LAN traffic and handles time-sensitive applications with reduced delays.

• Ensures standards compliant interoperability with RFC-1490 and SNMP management.

3.2.2 Network Topologies Supported

Multiplexer connected through satellite modems can supports three main types of satellite networks:

- Single star
- Distributed star
- Partial mesh
- Full mesh.

Below are diagram of linking multiplexers over full meshed topology VSAT network as shown in figure 3-3 and 3-4

3.2.2.1 Full Mesh Network

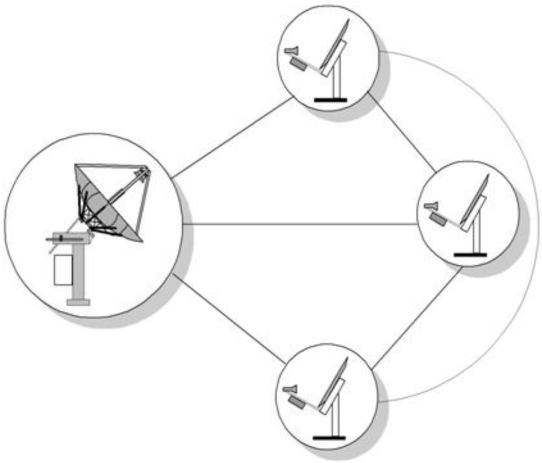


Figure 3-3: Full Mesh Network Connections

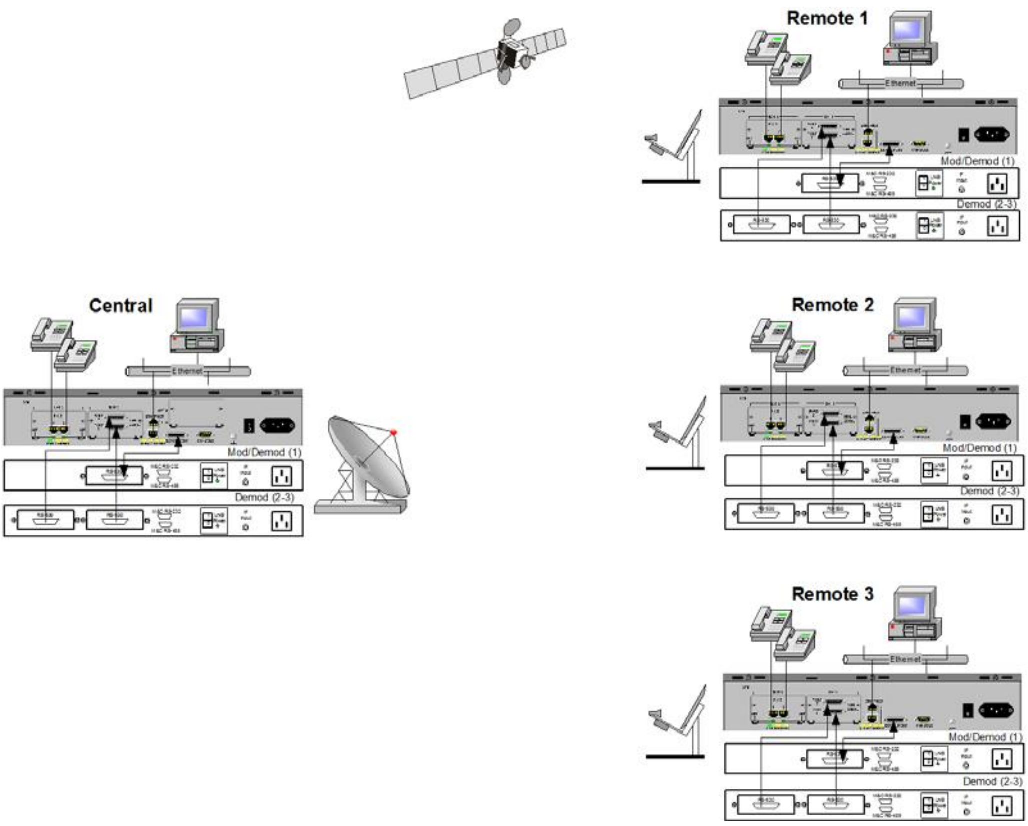


Figure 3-4: Full Mesh Network with multiplexers

3.2.3 Typical Multiplexer Over VSAT Applications

Multiplexer connected through VSAT network link can handle the following networks:

- Hybrid terrestrial/satellite networks with a mix of voice and data

• Networks that start small and need to grow. Satellite Modem and the Multiplexer provide modular expansion capabilities.

- Small to medium-sized enterprise networks:

- Star networks with up to 350 sites
- Distributed star networks with unlimited sites
- Mesh networks with up to 50 sites.

In particular, Multiplexer over VSAT is ideal for the following applications:

- Voice/data enterprise solutions via satellite
- E1/T1 voice trunking via satellite

3.2.3.1 Voice/data Enterprise Networks

In this application all multiplexer units in the enterprise network communicate via satellite connection.

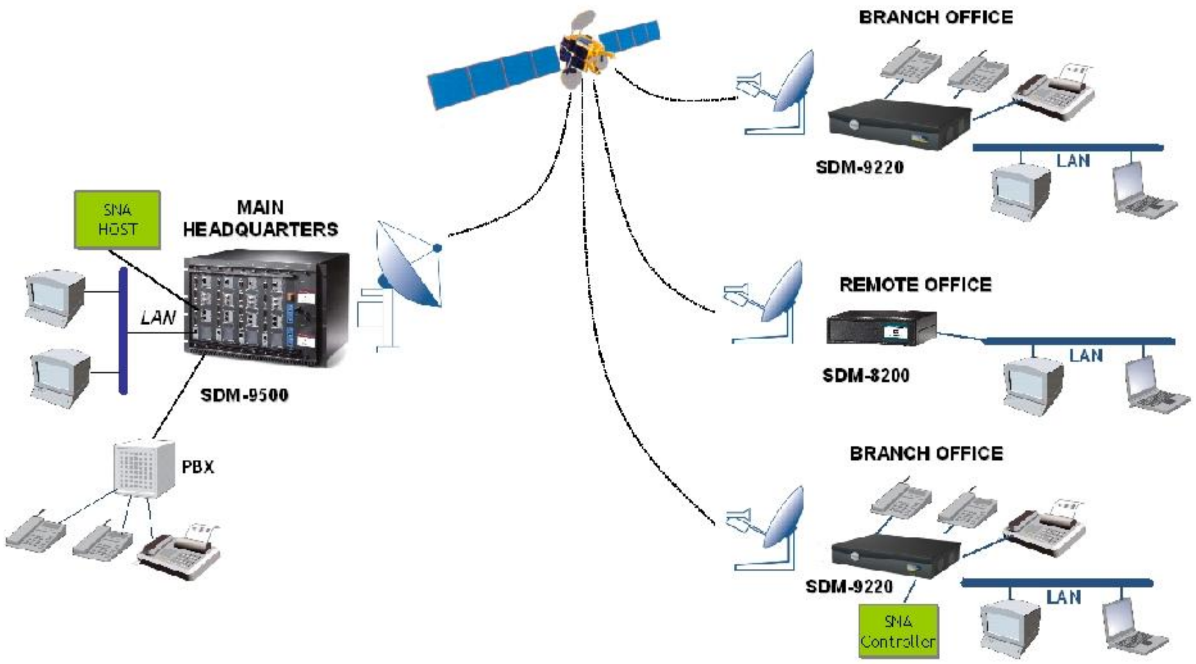


Figure 3-5: Voice/data Enterprise Network via Satellite

3.3 Services Protocols

3.3.1 Legacy Data Protocol

HDLC

HDLC (High-level Data Link Control) is a protocol defined by the International Standards

Organization. This protocol is a link-layer protocol having the following features:

• Error detection using a 16-bit, CRC-CCITT, cyclic redundancy checks error detection polynomial.

• Single frame format containing an arbitrary number of bits.

• Frame boundaries defined by a special bit pattern named FLAG (01111110).

• Data transparency by means of bit stuffing.

• Address field used to identify the station.

• Control field used for error correction and line establishment. The HDLC message format is:

01111110	ADDR	CTRL	DATA	CRC	01111110

The HDLC protocol is widely used, and many variations exist (ADCCP, transparent SDLC, X.25 level II, etc.). The variations differ only in their address and control field formats.

The Multiplexer’s HDLC driver is a transparent driver developed to support all HDLC- like protocols. It does not interpret or alter any bits in the ADDR, CTRL and DATA fields. This driver has the following characteristics:

The maximum frame size is 8200 bytes.

• Frame length should be a multiple of 8 bits, with a minimum length of 16 bits.

• Flags are stripped before transmission and regenerated at the remote end.

• Two coding modes: normal (NRZ) or NRZI.

• Two idle modes: FLAG or MARK.

The HDLC driver is used when a WAN/user port is configured with the HDLC driver:

• Performs flag and CRC stripping

• Integrally copies the X.25 into a Frame Relay frame (LAPB header and data),

• In the case of FRF.3 Annex F encapsulation, adds a Frame Relay header (2 bytes of DLCI) and a 6-byte frame identifier to the frame. [17][23]

3.3.2 IPX Connections

3.3.2.1 IPX Routing

The Multiplexer supports Internetwork Packet Exchange (IPX) routing for use in Novell networks. Packets are transported using the Ethernet 802.3, Ethernet V2 or Token-Ring 802.5 access protocols. The IPX layer handles addressing between networks and nodes, based on IPX network numbers.

Under IPX, the Multiplexer connects the various network segments and manages the addressing and routing of information packets. All network layer tasks are performed using RIP and the service Advertising Protocol (SAP). RIP is used to maintain the routing tables, and SAP is used to exchange service information between all routers in the network. [17]

3.3.2.2 The Static IP

If you disable IP RIP on the Multiplexer but require IP routing to a particular destination, you can configure a static IP address entry using the IP Static menu. Each entry is based on the IP address of the remote Multiplexer and the network address of the destination device.

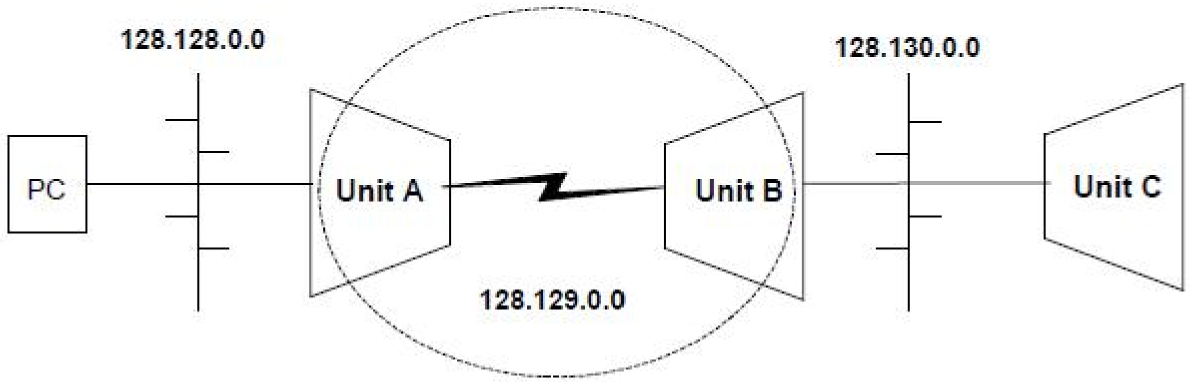


Figure 3-6 Examples of Static IP addresses

Table 3-1 Examples of Static IP addresses

Parameter	Local Unit	Remote Unit
IP Addr. of attached device:	128.128.0.1	128.130.0.1
IP Mask of attached device:	255.255.0.0	255.255.0.0
Default Gateway:	0.0.0.0	0.0.0.0
Default IP Address:	0.0.0.0	0.0.0.0
Default IP Mask:	0.0.0.0	0.0.0.0
LAN IP Address:	128.128.0.2	128.130.0.2
LAN IP Mask:	255.255.0.0	255.255.0.0
PVCR IP Address:	128.129.0.1	128.129.0.2
PVCR IP Mask:	255.255.0.0	255.255.0.0

The Setup IP Static menu lets you configure all parameters required to enable IP routing to a particular destination when IP RIP is disabled on the Multiplexer. For example, a PC attached to the local Multiplexer may be used for SNMP management of all devices attached to a remote LAN. To do this, the local unit must be able to route information across the remote LAN even if IP RIP is disabled.

3.3.3 Analog Voice

3.3.3.1 Signaling Engine Technology and Analog Voice Protocols

The Multiplexer Signalling Engine expansion board is equipped with:

• An MPC860 processor, responsible for handling the signalling used to establish the voice connection

• Communication interface cards, either digital or analog, which provide the physical connection to external devices

• State-of-the-art Digital Signal Processors (DSPs), which process the voice traffic using the following voice codec’s:

ACELP-CN: ACELP Comfort Noise (CN) at both 8 Kbps and 6 Kbps (refer to the next section). Available on both legacy Multiplexer products and the new Multiplexer product line, and can be used for interworking between the two.

ACELP-CN

The Multiplexer uses the ACELP (Algebraic Code Excited Linear Prediction) Comfort Noise (ACELP-CN) voice compression algorithm, or codec for superior throughput and voice quality.

ACELP-CN is a toll quality dual-rate codec that maintains high-quality sound with a compression rate of 8 Kbps or 6 Kbps.

It is ideal for multiplexing applications, can handle DTMF (Dual Tone Multi-Frequency) codes and provides a low-cost solution to maintaining voice quality in high-traffic networks.

It also offers bad/lost packet interpolation, reduced bandwidth during silence, a packet pace that permits double and triples buffering and improved quality for high-pitched voices.

3.3.3.2 Digital Signal Processor (DSP) Functions

Analog-to-digital (A/D) conversion must be performed before an analog voice signal can be carried over a digital line. A DSP is a microprocessor designed to digitize and process voice signals. The Multiplexer DSP carries out digitization and compression algorithms while consuming very little bandwidth. It is also used to handle other features of digitized voice processing, such as variable bit rates and echo cancelling.

Low-cost implementations of CELP-type compression algorithms in single-chip form became practical with the advent of the most recent generation of high-performance DSPs. The Multiplexer DSP and ACELP codec represent the latest advances in voice compression technology as shown in figure 3-7, and together provide very efficient voice compression.

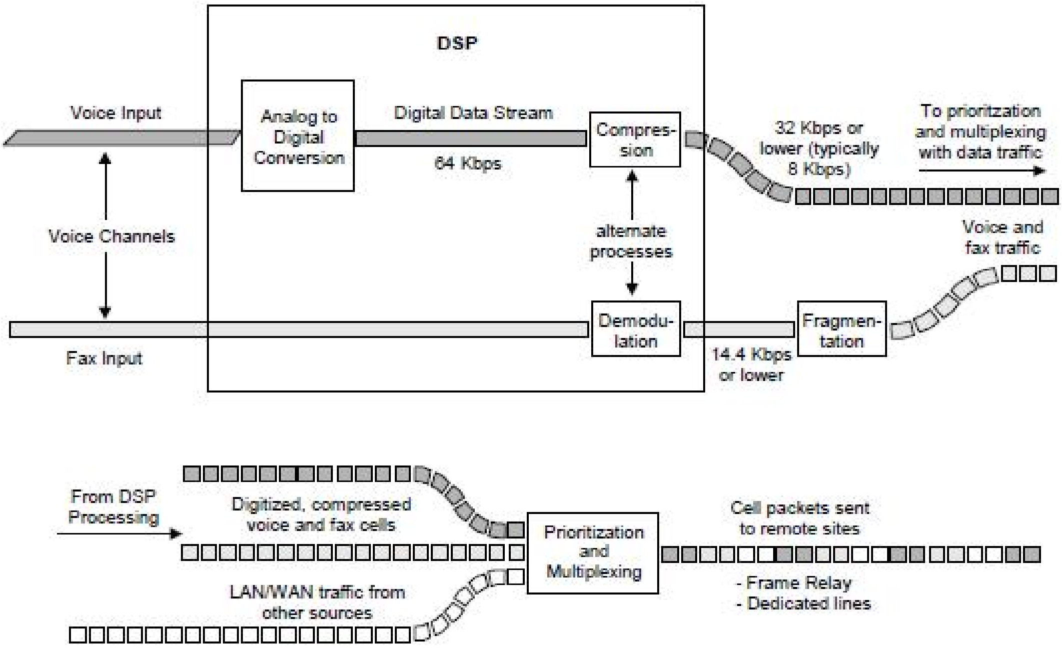


Figure 3-7: Processing of Voice/Fax Traffic

3.3.3.3 ACELP Compression/Decompression Procedure

• If the voice input is in analog format, the Multiplexer takes this analog source and converts it to a 64 Kbps digital stream in PCM (Pulse Code Modulation) for- mat.

• Using ACELP-CN, for example, the DSP cuts the data into 20 ms cells.

• The DSP then analyzes the voice spectrum and compresses the digital stream to 8 Kbps using the ACELP-CN algorithm. This provides a compression ratio of 8:1.

• The Multiplexer then combines the compressed voice cells with data from different sources according to assigned priorities. By default, voice traffic is defined as high priority, since it is extremely delay-sensitive.

• The mixed traffic is then transmitted over the wide area network using Multiplexer Cell Relay technology.

• At the remote end these processes are reversed. The remote channel's DSP receives the compressed voice traffic and decompresses it to a 64 Kbps digital PCM stream.

• If the output is analog, the remote Multiplexer reconverts the PCM stream to analog format and sends it to the attached voice equipment.

3.3.3.4 Fax Demodulation

When a Multiplexer DSP detects a fax tone it stops compressing the voice stream and starts demodulating the fax stream.

• The Multiplexer demodulates the fax signals into HDLC (High-level Data Link

Control) data at speeds of 14.4 Kbps or lower.

• The HDLC data is then fragmented into cells.

• The Multiplexer then combines the HDLC/fax cells with data from different sources according to assigned priorities. Fax, which is extremely delay-sensitive, is given high-priority status by default.

• The mixed traffic is then transmitted to remote sites using Cell Relay technology.

• At the remote end these processes are reversed. The remote channel receives the

HDLC/fax cells and converts them to digital fax signals.

• The remote Multiplexer then reconverts the digital stream to analog format and sends it to the attached fax equipment.

Since audio transmission signals at 64 Kbps are converted to a digitized stream at 14.4

Kbps or lower, fax demodulation reduces the required bandwidth. Combined with the advantages of fragmentation and cell relay, the result is more efficient transport of fax signals, with reduced delays.

3.3.3.5 Variable Bit Rate

The Multiplexer’s ACELP-CN compression algorithm produces 8 Kbps or 6 Kbps voice output. However, the Multiplexer can lower the bit rate even further depending on the nature of the voice stream. The speed is automatically reduced to a lower bit rate when:

• Signalling and DTMF tones are transmitted, or

• Silent periods occur. The reduced bandwidth is used to maintain background noise on the line (without some background noise, users perceive the line to be dead).

Voice communication is intrinsically half-duplex by nature: when one person speaks at one end of the line, the person at the other end listens. Pauses may also occur during speech, for example, between sentences, when the speaker leaves the phone, or when the speaker puts the listener on hold. The Multiplexer detects these silence periods. Its variable bit rate takes advantage of the fact that noise and DTMF signalling tones require less bandwidth than the voice traffic itself. It can then allocate the bandwidth saved from a silent or signalling voice channel to channels that are processing voice traffic. The result of a variable bit rate is optimized bandwidth utilization, improved system performance and a reduction of overall network costs.

3.3.3.6 Echo Cancelling

Echo is caused by impedance mismatches on the telephone circuit. It is a distorted and delayed replica of the incoming speech from the remote end.

The Multiplexer has an echo canceller built into its DSP. Should echo occur on a speech or non-speech signal (such as voice-band data or fax) the echo cancelled automatically reduces this echo to tolerable levels. It adapts automatically to changes in the echo that may occur in successive connections along the virtual path. It also minimizes background noise and prevents the negative effects of double talk. Echo cancellation can be enabled or disabled using the Echo Canceller parameter. By default, it is disabled.

3.3.4 Analog Voice Connections

A Multiplexer analog port connects directly to an analog telephone.

- An FXS interface card is used when connecting to an analog telephone or KTS unit

3.3.4.1 E&M Interface Card

• The E&M interface can use a two- or four-wire circuit. Type I, II and V are sup- ported on the Multiplexer

• The number of E&M lines supported varies from 1 to 16, depending on the multiplexer model, the number of interface cards installed and number of DSPs available

• E&M uses conventional analog station interfaces which adhere to EIA/TIA Voice band and Loop Signalling application standards.

3.3.4.2 Signaling Variations Supported

The E&M interface card supports the following signalling variations:

- Immediate Start: An E&M type where transmission takes place immediately.

This is the industry standard for E&M operation, and the default setting on the card

- Wink Start: An E&M type where the unit toggles the A/B-lead before the PBX

Will transmit dial digits

• Custom: Custom signalling settings that you can use to fine-tune your application. FXS Interface Card

• An FXS interface card connects a telephone or KTS unit directly to the multiplexer unit, using the FXS voice protocol

• The Multiplexer presents a Telco/PTT interface that acts like a Central Office and can interface to a conventional two-wire telephone (pulse-dial or touch-tone)

• The numbers of FXS connections supported can vary from 1 to 8, depending on the multiplexer model, the number of interface cards installed and number of DSPs available

• FXS uses the Loop Start Signalling method to seize and sense a line

Loop Start Signalling uses 2 wires, Tip and Ring, to perform signalling and carry Voice Frequency (VF) signals.

A relay opens or closes the loop between a particular subscriber and the multiplexer FXS port. This generates current flow into the loop, which is detected by the switching equipment.

• A Multiplexer FXS port provides loop current and ring voltage, and detects the off-hook and on-hook states.

3.3.5 Tones Generated by the Multiplexer

The following tones can be generated by the Multiplexer for analog voice calls:

• Audio tones: Dial tone, ringback tone and busy signals. These are generated according to North American standards. Thus if you make a call from London, you will hear the North American signalling tones, not those for the United King- dom. Busy signals include:

- Slow busy, generated when the destination is busy

- Fast busy, generated when the link goes down

- Incompatibility tone, generated when some fatal problem with the voice connection occurs, for example, the voice algorithms is mismatched.

• Physical tone: Remote (equipment) ring on the telephone set, generated from the electrical signal originating from an attached CO. The frequency at which the Multiplexer generates a ring is governed by the global Ring frequency parameter, and may be 17, 20, 25 or 50 Hz. Its voltage is governed by the global Ring voltage parameter, and may be 60 or 80 Volts RMS.

• Multi-frequency tones: These include the DTMF, MF and R2 tones. Their signals are passed transparently when a conversation is in progress. They have no effect at any time for predefined line activation (see next section). For switched line activation, they are intercepted in the early stages of the calling procedure to determine the destination unit and port from the Voice Mapping Table, and are then passed transparently once the call is placed. They can be used for interactive touch tone procedures during a call.

3.3.6 Line Activation Types

For full network flexibility, voice/fax line activation can be configured as switched, predefined, autodial or broadcast. Use the Activation type parameter on the voice channel, which can be set as SWITCHED, PREDEFINED, AUTODIAL or BROADCAST

3.3.6.1 Predefined Line Activation

For predefined activation, the destination unit and port number are preconfigured by the user. As soon as the device connected to the local port goes off-hook, the local Multiplexer begins the calling procedure with the destination device. In other words, all you have to do is lift the telephone receiver and the remote telephone will ring immediately. This configuration can be used only when a dedicated telephone is available at both the source and destination sites. Often, a more popular alternative is the autodial connection.

The two voice ports linked through predefined line activation cannot be accessed by any other voice port in the network.

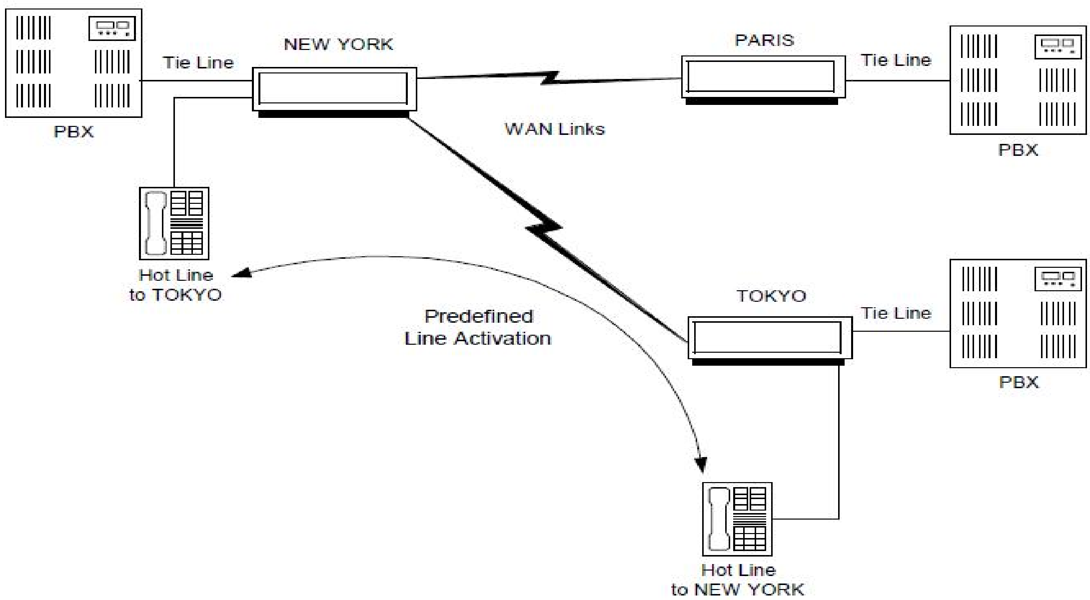


Figure 3-8: Predefined Line Activation

3.3.6.2 Switched Line Activation

Under switched activation (voice channel Activation type parameter set to SWITCHED), the Multiplexer selects the remote location according to a configurable Speed Dial Number that the user enters into the telephone set.

- No predetermined connection is set up between any two voice channels

- All speed dial numbers are kept in a Voice Mapping Table along with the associated destination unit, optional extension number, extension number length, and an optional dialling sequence (extended digits) that can be forwarded to the attached voice equipment.

- Speed dial numbers in the Voice Mapping Table are variable in size (from 1 to 30 digits). The Multiplexer determines that a dialling sequence is completed when the global (inter-digits) Dial timer expires or when the user terminates dialling with the pound sign (#).

The advantages of switched line activation are:

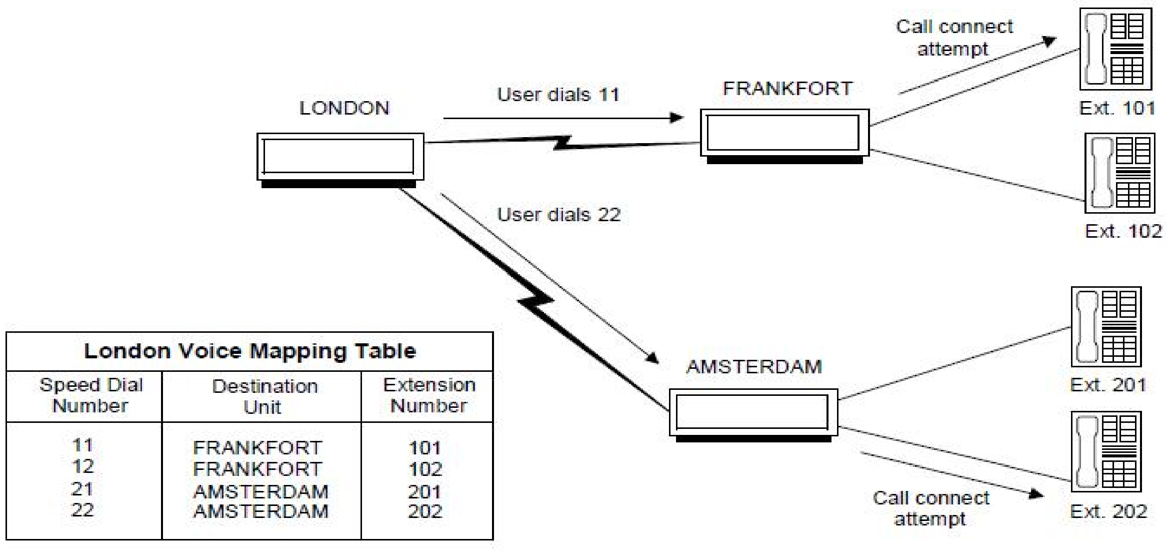
- You can have an unequal number of channels at different sites. This provides a more practical and cost-effective approach to voice networks.

Three or more offices can be interconnected without requiring a central PBX or multiple compression/decompression cycles. This method of voice/fax switching reduces delays and requires less bandwidth than for a central PBX setup.

With switched line activation, the multiplexer base product also supports:

Local voice switching: Permits a switched voice call from one voice channel to another on the same unit

Hunt Groups: Permits attempting more than one voice channel when trying to place an ingress call.



London Voice Mapping Table		
Speed Dial Number	Destination Unit	Extension Number
11	FRANKFORT	101
12	FRANKFORT	102
21	AMSTERDAM	201
22	AMSTERDAM	202

Figure 3-9: Switched Line Activation

3.3.6.3 Specialized Line Activation

The Multiplexer supports analog voice applications designed for special circumstances:

• Push To Talk used for ground-to-air radio communications such as air traffic control, or communications between two remote locations. The audio path is always up.

3.3.6.4 Push To Talk

Hardware Support

The Multiplexer supports a Push To Talk (PTT) application on the E&M interface card.

• Supports applications that cannot use the signalling types available on the E&M interface to carry the PTT signals

• Uses +24 VDC rather than -48VDC for these applications

• Eliminates the need for an expensive PTT to E&M converter.

Contact Multiplexer Technical Support for information on the availability of this new card. Be prepared to provide details on the PTT signal levels required for your application.

Operations

PTT is used to key a radio in two different scenarios:

• Scenario: From a remote mobile location to a control station, such as an air traffic control application.

. In of this scenario:

• The audio path is always up

• The PTT CONTROL device produces M lead transitions on the E&M connection

• The Multiplexer detects these M lead transitions, and regenerates them as E lead transitions

• The E lead transitions are transmitted and replayed at the other end of the connection without clearing the audio path

• As a result, more rapid communications can take place. In an application involving a fixed control station (Scenario 1):

• The M lead is permanently connected to Signal Ground at the PTT ANSWER side, so that the E&M channel is up at all times

• Only the control station operates as a PTT CONTROL device. In an application involving two remote locations (Scenario 2):

• The M lead is not permanently connected to Signal Ground at either end of the connection

• The E&M connection goes up and stays up once the first M lead transition occurs, in other words, when the first user at either location presses the talk but- ton

• Users at both locations can use their radio as a PTT CONTROL device, producing transitions on the M lead which the Multiplexer detects and regenerates as E lead transitions at the opposite end of the connection.

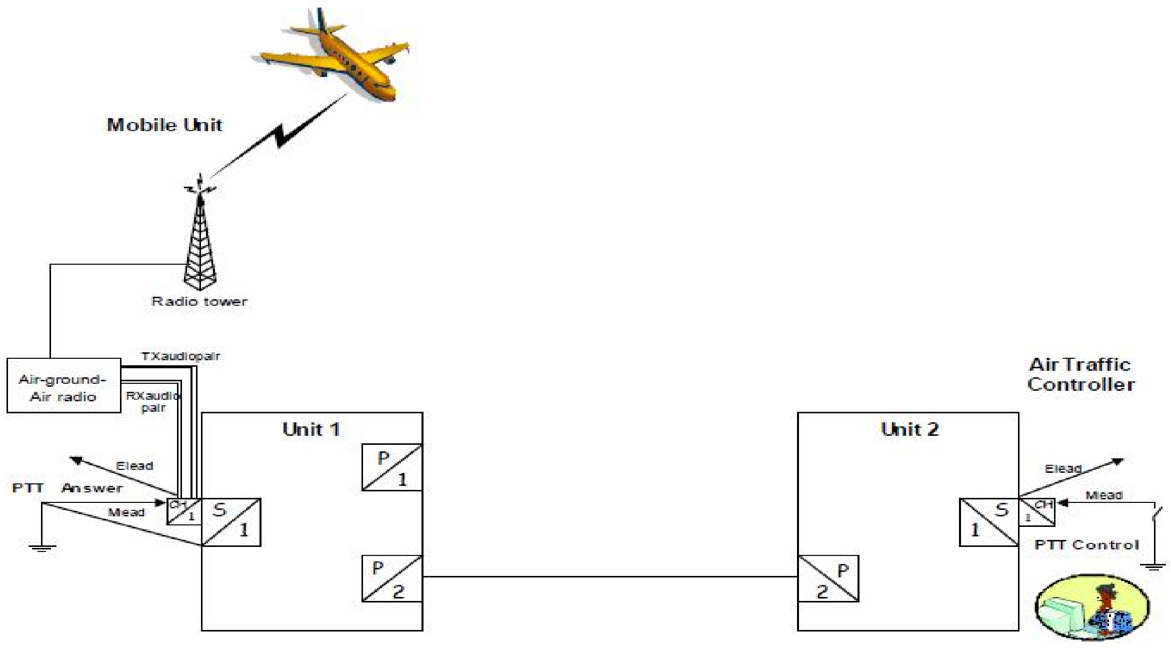


Figure 3-10: Push-To-Talk Application: Scenario

CHAPTER FOUR

Models Configuration and Results

In this chapter, we present the results of the implementation of The Models described in Chapter 3 the implemented using the concepts and models presented in Chapters 2and 3. The implementation provides results for performance of E1 Model link solution and IP VSAT Model solution. Results are provided for the performance for both network Models.

4.1 E1 Model Setup and Configuration

The figure below shows three multiplexers connecting together through E1 cards network, specifing the each unit name and WAN and Channel port numbers used to communicate to each others

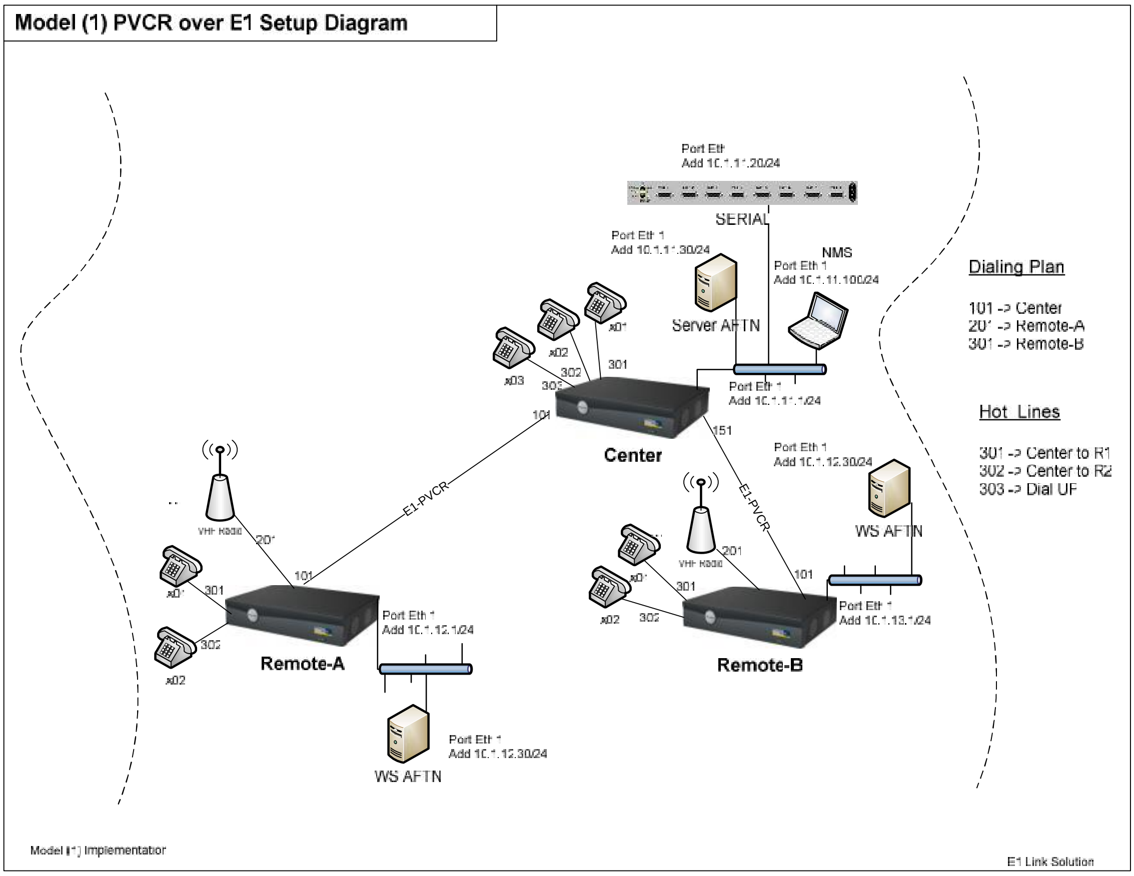


Figure 4-1 E1 Model PVCR over E1 Setup Diagram

E1 Model Configuration Steps

4.1.1 CENTER Site – Parameters Configuration

- CENTER: Global Parameters
- CENTER: Port 100 Parameters (WAN) – SE/SLOT/1/PORT 1/ LINK
- CENTER: Port 101 Parameters (WAN) – SE/SLOT/1/PORT 1/CH 101
- CENTER: Port 150 Parameters (WAN) – SE/SLOT/1/PORT 2/ LINK
- CENTER: Port 151 Parameters (WAN) – SE/SLOT/1/PORT 2/CH 151
- CENTER: E&M port 200 (Analog Voice) – SE/SLOT/2/LINK
- CENTER: E&M ports 201-202 (Analog Voice) – SE/SLOT/2/CH 1 & 2
- CENTER: FXS port 300 (Analog Voice) – SE/SLOT/3/LINK
- CENTER: FXS ports 301-302 -303 (Analog Voice) – SE/SLOT/3/CH 1 & 2 & 3
- CENTER: Eth 2 Parameters (User IP) – SE/PORT/ETH2
- CENTER: IP Global Parameters – SE/IP/GLOBAL
- CENTER: IP HTTP Parameters – SE/IP/HTTP
- CENTER: PVC 1Parameters (FR over IP) – SE/PVC/1
- CENTER: IP STATIC– SE/IP/STATIC
- CENTER: MAP FILE– SE/MAP/ADD

4.1.2 CENTER Site SERIAL Unit Parameters Configuration

- SERIAL: Global Parameters – SE/GLOBAL
- SERIAL: Eth Parameters (User IP) – SE/PORT/ETH
- SERIAL: IP Global Parameters – SE/IP/GLOBAL
- SERIAL: IP HTTP Parameters – SE/IP/HTTP
- SERIAL: PVC 1Parameters (FR over IP) – SE/PVC/1
- SERIAL: Port 1&2 Parameters (USER DATA) – SE/PORT/1&2

4.1.3 REMOTE-A: Unit Parameters Configuration

- REMOTE-A: Global Parameters – SE/GLOBAL
- REMOTE-A: Port 100 Parameters (WAN) – SE/SLOT/1/PORT 1/ LINK
- REMOTE-A: Port 101 Parameters (WAN) – SE/SLOT/1/PORT 1/CH 101
- REMOTE-A: E&M port 200 (Analog Voice) – SE/SLOT/2/LINK
- REMOTE-A: E&M ports 201 (Analog Voice) – SE/SLOT/2/CH 1
- REMOTE-A: FXS port 300 (Analog Voice) – SE/SLOT/3/LINK
- REMOTE-A: FXS ports 301-302 (Analog Voice) – SE/SLOT/3/CH 1 & 2
- REMOTE-A: Eth 2 Parameters (User IP) – SE/PORT/ETH2
- REMOTE-A: IP Global Parameters – SE/IP/GLOBAL
- REMOTE-A: IP HTTP Parameters – SE/IP/HTTP
- REMOTE-A: PORT 1 Parameters (User DATA) – SE/PORT/1
- REMOTE-A: MAP FILE– SE/MAP/ADD

4.1.4 REMOTE-B: Unit Parameters Configuration

- REMOTE-B: Global Parameters – SE/GLOBAL
- REMOTE-B: Port 100 Parameters (WAN) – SE/SLOT/1/PORT 1/ LINK
- REMOTE-B: Port 101 Parameters (WAN) – SE/SLOT/1/PORT 1/CH 101
- REMOTE-B: E&M port 200 (Analog Voice) – SE/SLOT/2/LINK
- REMOTE-B: E&M ports 201 (Analog Voice) – SE/SLOT/2/CH 1
- REMOTE-B: FXS port 300 (Analog Voice) – SE/SLOT/3/LINK
- REMOTE-B: FXS ports 301-302 (Analog Voice) – SE/SLOT/3/CH 1 & 2
- REMOTE-B: Eth 2 Parameters (User IP) – SE/PORT/ETH2
- REMOTE-B: IP Global Parameters – SE/IP/GLOBAL
- REMOTE-B: IP HTTP Parameters – SE/IP/HTTP
- REMOTE-B: PORT 1 Parameters (User DATA) – SE/PORT/1
- REMOTE-B: MAP FILE– SE/MAP/ADD

4.2 VSAT Model Setup and Configuration

The figure below shows three multiplexers connecting together through full meshed VSAT network, specifing the each unit name and IP address and their DLCI numbers used to communicate to each others

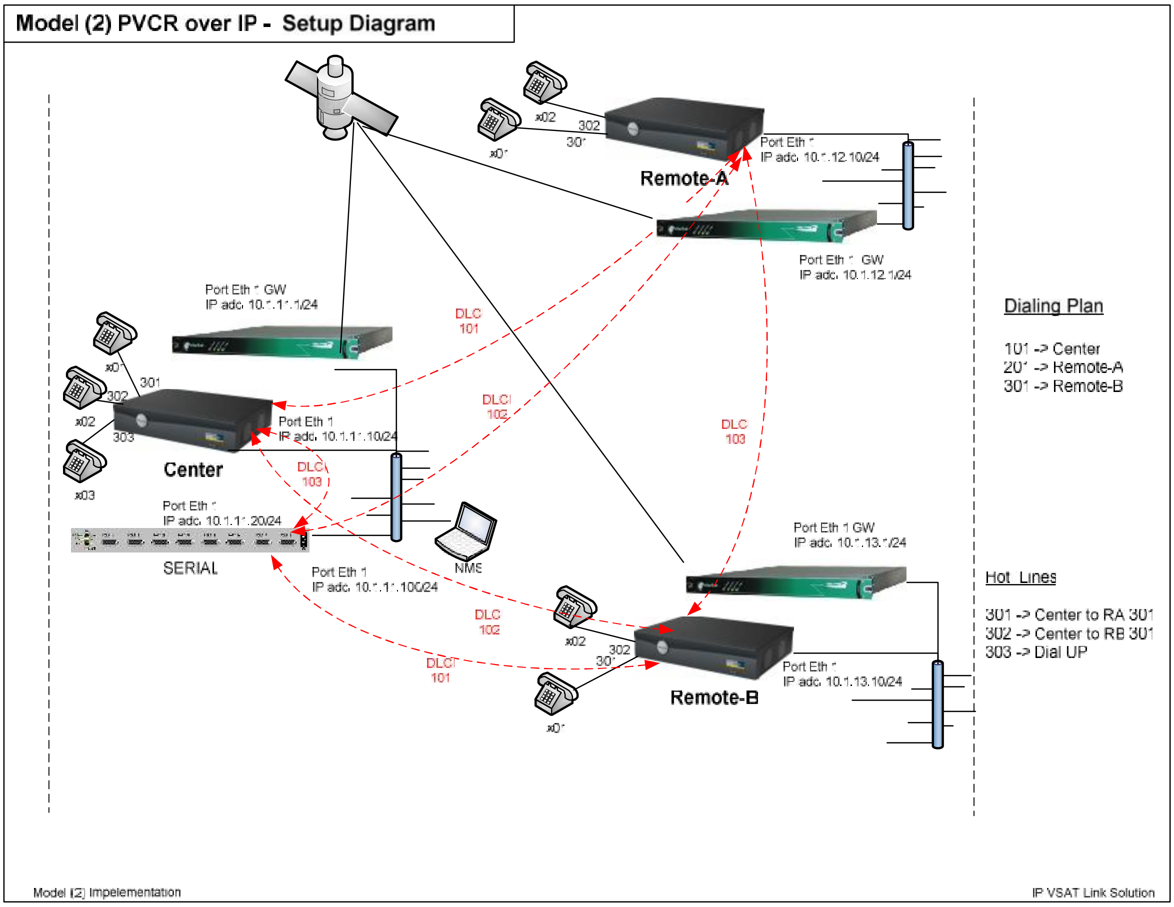


Figure 4-2 VSAT Model PVCR over IP VSAT Setup Diagram

4.2.1 CENTER: Unit Configuration – Central Site

- CENTER: Global Parameters – SE/GLOBAL
- CENTER: Eth 2 Parameters (User IP) – SE/PORT/ETH2
- CENTER: PVC 1, 2 & 3 Parameters (FR over IP) – SE/PVC/1, 2 & 3

4.2.2 SERIAL: Unit Configuration – Central Site

- SERIAL: Global Parameters – SE/GLOBAL
- SERIAL: PVC 1, 2 & 3 Parameters (FR over IP) – SE/PVC/1, 2 & 3

4.2.3 REMOTE-A: Unit Configuration – Remote Site 1

- REMOTE-A: Global Parameters – SE/GLOBAL
- REMOTE-A: Eth 2 Parameters (User IP) – SE/PORT/ETH2
- REMOTE-A: PVC 1, 2 & 3 Parameters (FR over IP) – SE/PVC/1, 2 & 3

4.2.4 REMOTE-B: Unit Configuration – Remote Site 2

- REMOTE-B: Global Parameters – SE/GLOBAL
- REMOTE-B: Eth 2 Parameters (User IP) – SE/PORT/ETH2
- REMOTE-B PVC 1, 2 & 3 Parameters (FR over IP) – SE/PVC/1, 2 & 3

4.2.5 VSAT Network Setup and Configuration

4.2.5.1 Network and Terminal

• Configure network parameters:

– Network size;

– Frame time, 40ms by default,

– Symbol rate;

– Number of frames per super frame;

– Number of CSC burst per frame;

– Satellite longitude;

– Signalling carrier frequency,

• Configure terminal parameters

– Local node number

– Signalling frequency

– Nominal TX power

• Maximum TX power

• Nominal RX level

– TCP Accelerator Disable/Enable

• Accelerator

• Compression

4.2.5.2 IP address and Routing

• Configure Route

– Local IP address/mask for each terminal in the network.

• It also defines the subnet that is accessible from Ethernet port.

– Gateway or the route for the terminal

– Only the primary reference terminal is allowed to configure the route

4.2.5.3 Bandwidth and BoD

• Configurable BW assignment

– BW assignment is link based;

• Min – Minimum BW; CIR – Committed BW; EIR – Excess BW;

– A dedicated link setting is allowed for each connection from one terminal to another;

– The default link BW parameters setting is applied to all links if there is no dedicated settings for BW,

– Assignment is based on the request from user traffic.

– Pre-configured link is allowed to set source and destination terminal; one direction;

– Bi-direction link needs to configure two links, it could be asymmetrical;

4.2.5.4 TDMA Carrier(s)

• Configure carrier

– Show signalling carrier frequency defined from terminal configuration;

– Up to 32 carriers can be configured.

– Only the primary reference terminal is allowed to configure the carrier

4.3 Models Practical Configuration

4.3.1 E1 Model Practical Configuration Results

This figure below shows:

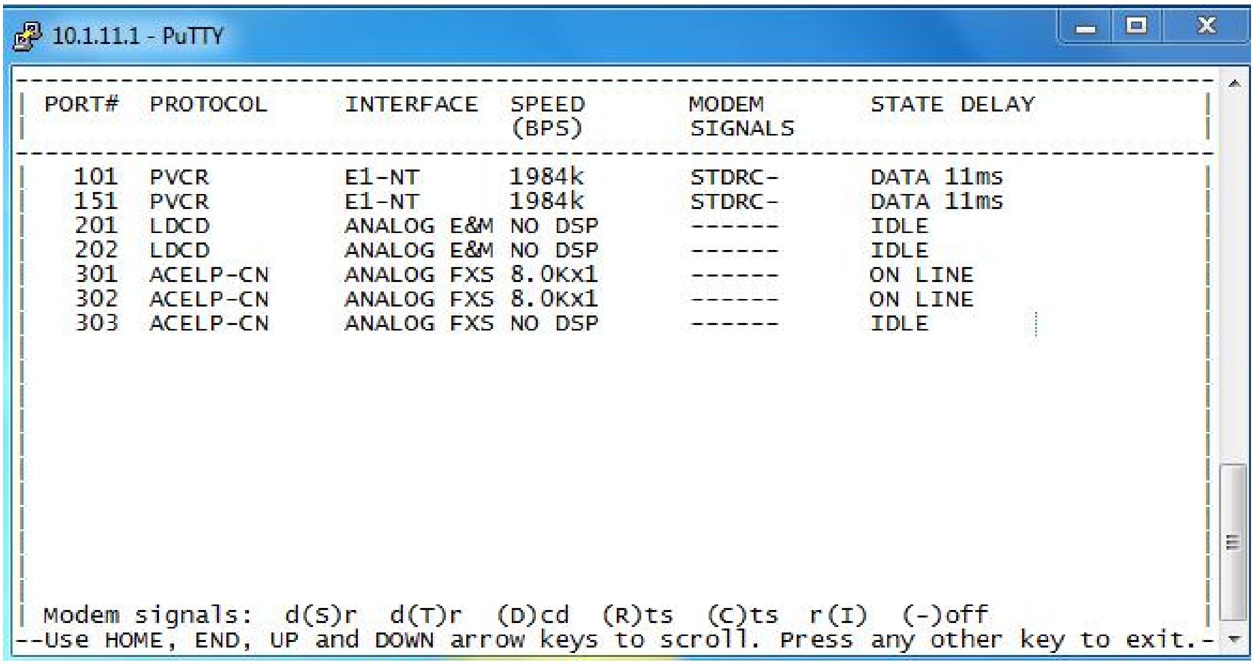
- Photo displaying Practical setup and configuration of three multiplexers connecting to each other through E1 Link Cards
- The telephones machine for testing (Telephones and VHF Lines)
- Data cables for connecting (Radars Services)
- PC for configuration and testing IP Traffic connectivity form all stations

Figure 4-3: Using E1 Cards Links the Multiplexers

A. WAN (E1) and Voice (VHF + Hotline + Dialup) ports Connectivity

The figure below shows:

- Capture screen from CENTER multiplexer displaying:
- WAN ports Connectivity from CENTER Station to the REMOTE Stations (on DATA mean the CENTER Site online with two other station with a delay of 11ms)
- CENTER Site Voice Ports connectivity to REMOTE Sites ,IDLE (mean no Traffic but the port is configured , On Line (mean there are phones call for Center to Remote)


10.1.11.1 - PuTTY

PORT#	PROTOCOL	INTERFACE	SPEED (BPS)	MODEM SIGNALS	STATE	DELAY
101	PVCR	E1-NT	1984k	STDRC-	DATA	11ms
151	PVCR	E1-NT	1984k	STDRC-	DATA	11ms
201	LDCD	ANALOG E&M	NO DSP	------	IDLE	
202	LDCD	ANALOG E&M	NO DSP	------	IDLE	
301	ACELP-CN	ANALOG FXS	8.0Kx1	------	ON LINE	
302	ACELP-CN	ANALOG FXS	8.0Kx1	------	ON LINE	
303	ACELP-CN	ANALOG FXS	NO DSP	------	IDLE	

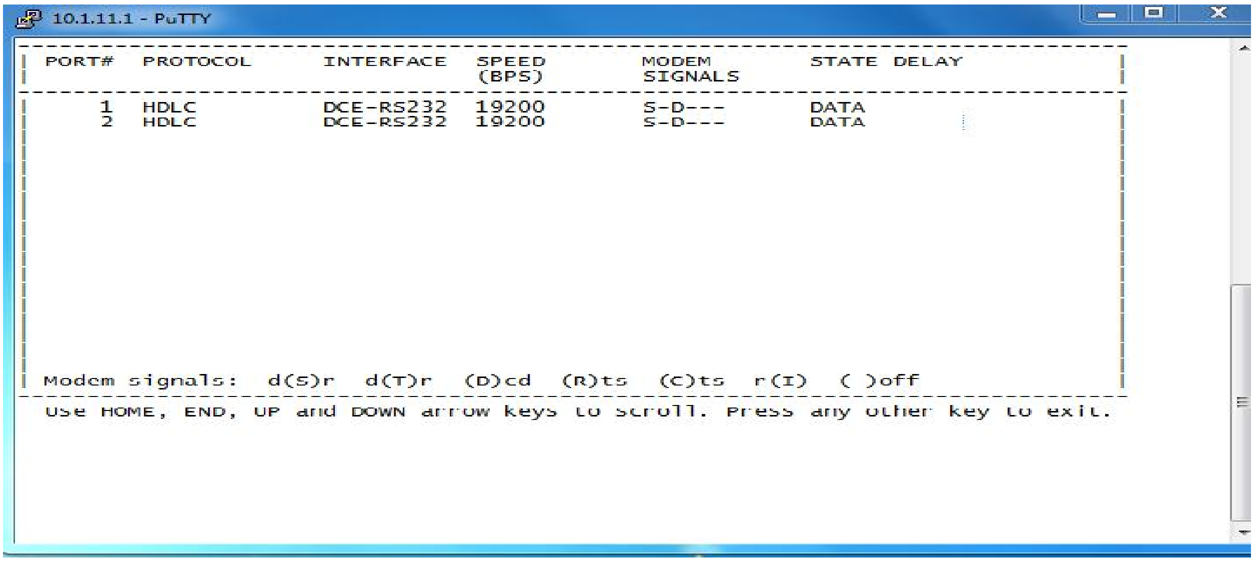
Modem signals: d(S)r d(T)r (D)cd (R)ts (C)ts r(I) (-)off
--Use HOME, END, UP and DOWN arrow keys to scroll. Press any other key to exit.-

Figure 4-4: CENTER to REMOTE Stations Voice Ports and E1 Link Connectivity

B. Data Ports (Radars) Connectivity

The figure below shows:

- Capture screen from CENTER multiplexer displaying two radars ports connection between the center site and the two remote sites (DATA means the port is online and ready for sending traffic over it)


10.1.11.1 - PuTTY

PORT#	PROTOCOL	INTERFACE	SPEED (BPS)	MODEM SIGNALS	STATE	DELAY
1	HDLC	DCE-RS232	19200	S-D---	DATA	
2	HDLC	DCE-RS232	19200	S-D---	DATA	

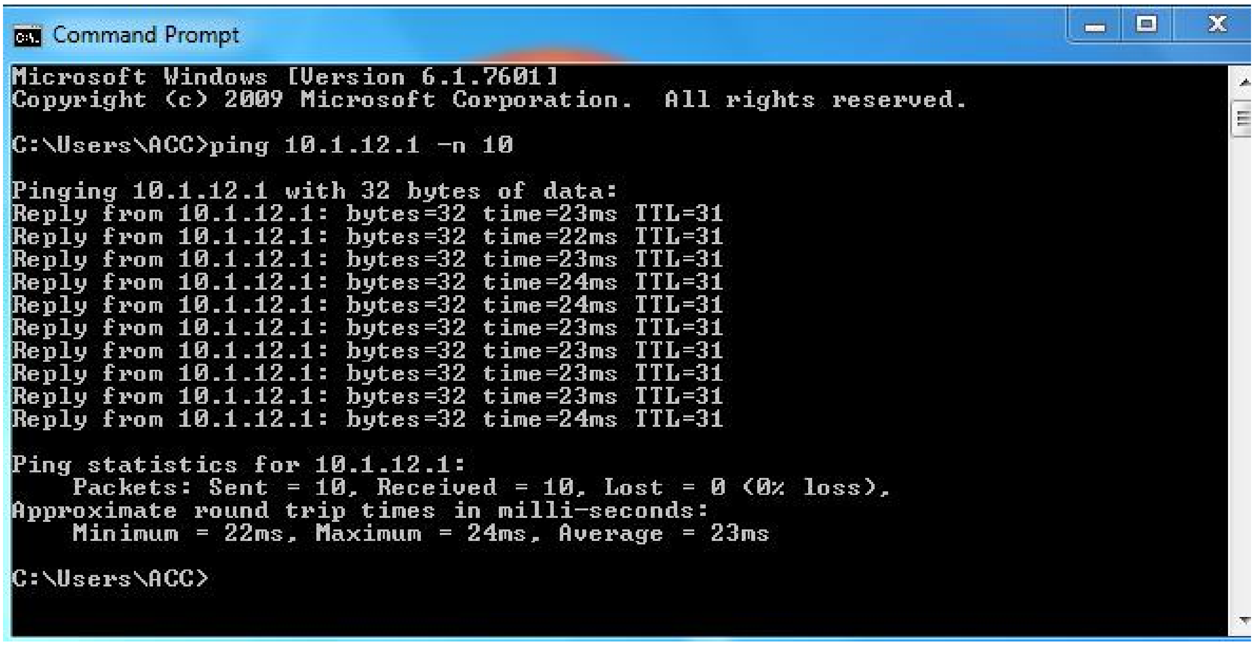
Modem signals: d(S)r d(T)r (D)cd (R)ts (C)ts r(I) ( )off
Use HOME, END, UP and DOWN arrow keys to scroll. Press any other key to exit.

Figure 4-5: CENTER Site (Serial) to REMOTE Sites Data Ports Connectivity

C. IP Connectivity

The figure below shows:

- Capture screen from CENTER NMS PC displaying IP Connectivity between **CENTER** Station and **REMOTE-A** Station for IP User Traffic

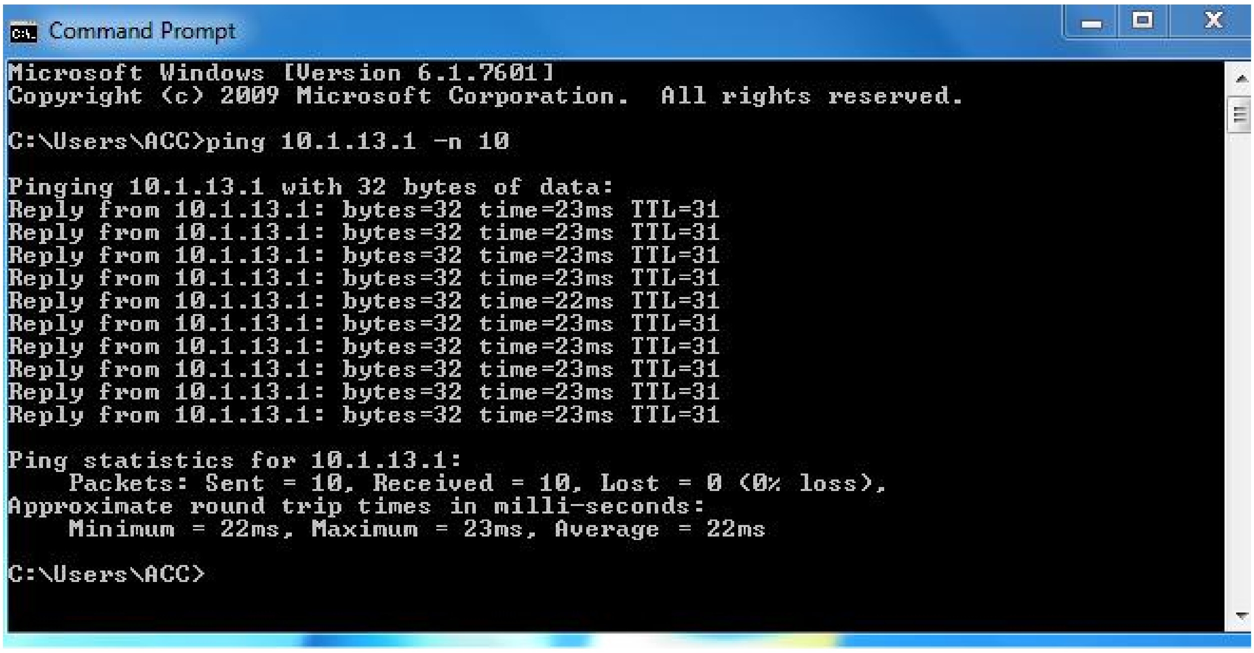


```
Command Prompt
Microsoft Windows [Version 6.1.7601]
Copyright (c) 2009 Microsoft Corporation.  All rights reserved.

C:\Users\ACC>ping 10.1.12.1 -n 10

Pinging 10.1.12.1 with 32 bytes of data:
Reply from 10.1.12.1: bytes=32 time=23ms TTL=31
Reply from 10.1.12.1: bytes=32 time=22ms TTL=31
Reply from 10.1.12.1: bytes=32 time=23ms TTL=31
Reply from 10.1.12.1: bytes=32 time=24ms TTL=31
Reply from 10.1.12.1: bytes=32 time=24ms TTL=31
Reply from 10.1.12.1: bytes=32 time=23ms TTL=31
Reply from 10.1.12.1: bytes=32 time=23ms TTL=31
Reply from 10.1.12.1: bytes=32 time=23ms TTL=31
Reply from 10.1.12.1: bytes=32 time=23ms TTL=31
Reply from 10.1.12.1: bytes=32 time=24ms TTL=31

Ping statistics for 10.1.12.1:
    Packets: Sent = 10, Received = 10, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 22ms, Maximum = 24ms, Average = 23ms

C:\Users\ACC>
```

Figure 4-6: CENTER Site to REMOTE-A IP Connectivity

The figure below shows:

- Capture screen from CENTER NMS PC displaying IP Connectivity between **CENTER** Station and **REMOTE-B** Station for IP User Traffic



```
Command Prompt
Microsoft Windows [Version 6.1.7601]
Copyright (c) 2009 Microsoft Corporation.  All rights reserved.

C:\Users\ACC>ping 10.1.13.1 -n 10

Pinging 10.1.13.1 with 32 bytes of data:
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=22ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31
Reply from 10.1.13.1: bytes=32 time=23ms TTL=31

Ping statistics for 10.1.13.1:
    Packets: Sent = 10, Received = 10, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 22ms, Maximum = 23ms, Average = 22ms

C:\Users\ACC>
```

Figure 4-7: CENTER Site to REMOTE-B IP Connectivity

4.3.2 VSAT Model Practical Configuration Results

The two figure above shows:

- Photos displaying Practical setup and configuration of three multiplexers connecting to each other through Three VSAT Modems
- The Satellite modems connecting together in full mesh VSAT Network using Splitter for RX Signals and Combiner for TX signals
- Multiplexer station connect to VSAT Modem using IP Ethernet Connection

Figure 4-8: Full Mesh VSAT Network using three Modems to connect Multiplexers

VSAT Stations Connectivity

The figure below shows:

- Capture screen from CENTER (Primary Station) displaying all three VSAT Station are Online and communicating each other

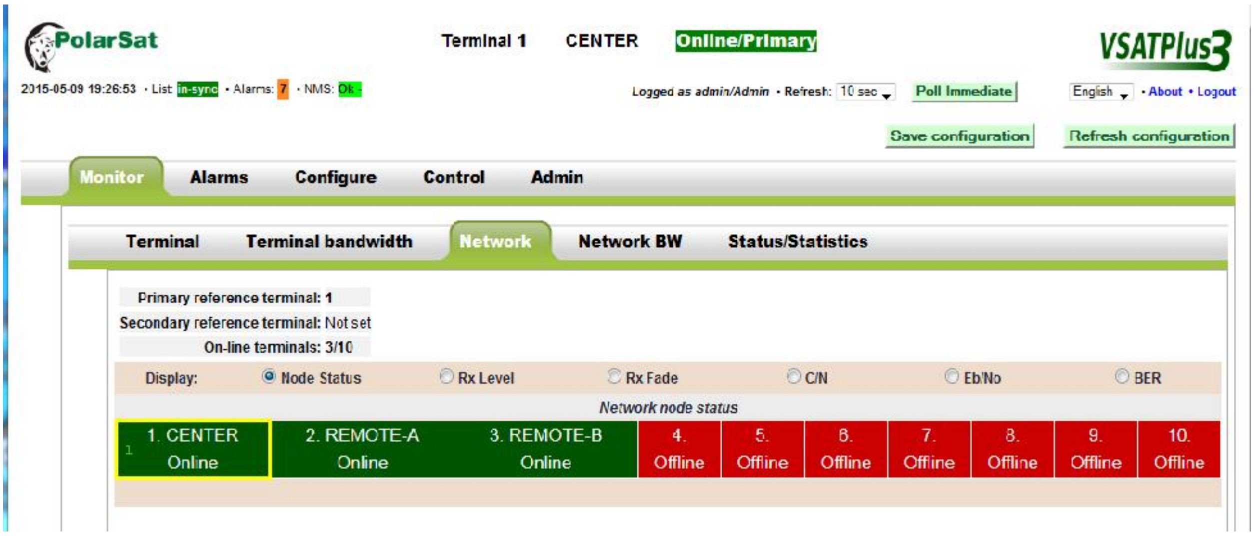


Figure 4-9: Three VSAT Stations Networks

A. CENTER Station Terminal Parameters Status

The figure below shows:

- Capture screen from CENTER (Primary Station) displaying its own Status showing that the station is ONLINE and IN SYNCH.

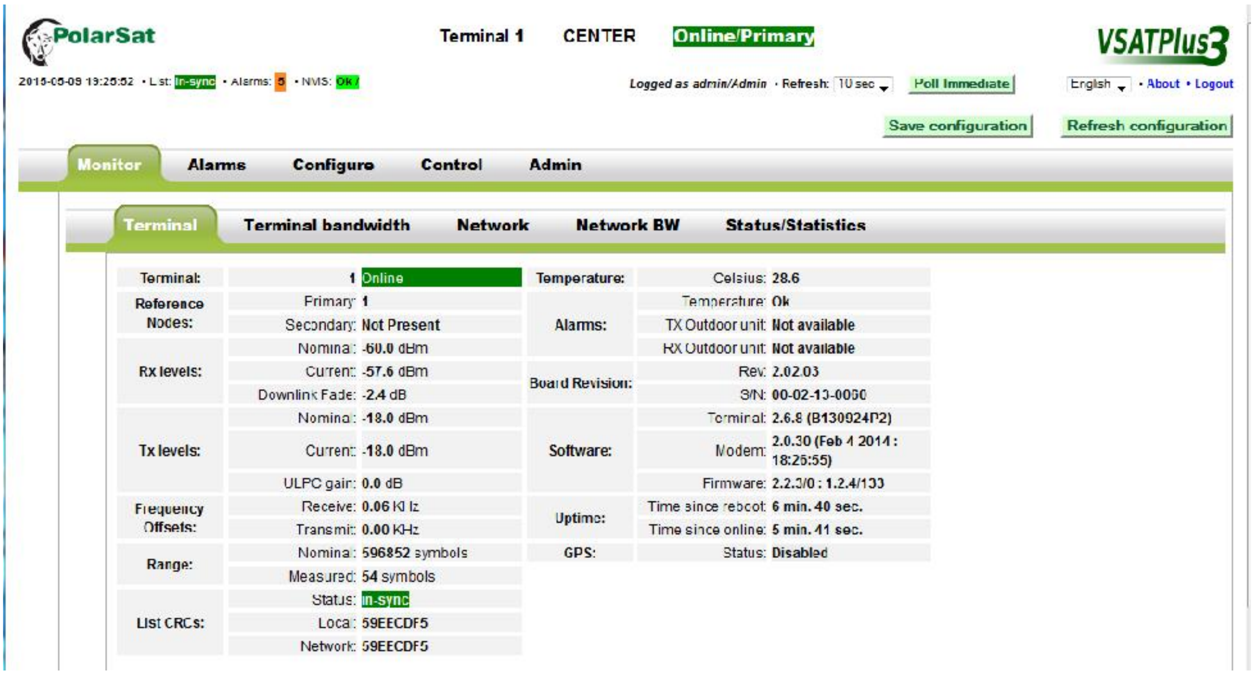


Figure 4-10: CENTER Terminal Parameters Status

Network Statics IP Routing Configuration (only done once at primary station)

The figure below shows:

- Capture screen from CENTER (Primary Station) showing IP Routing between the three Stations

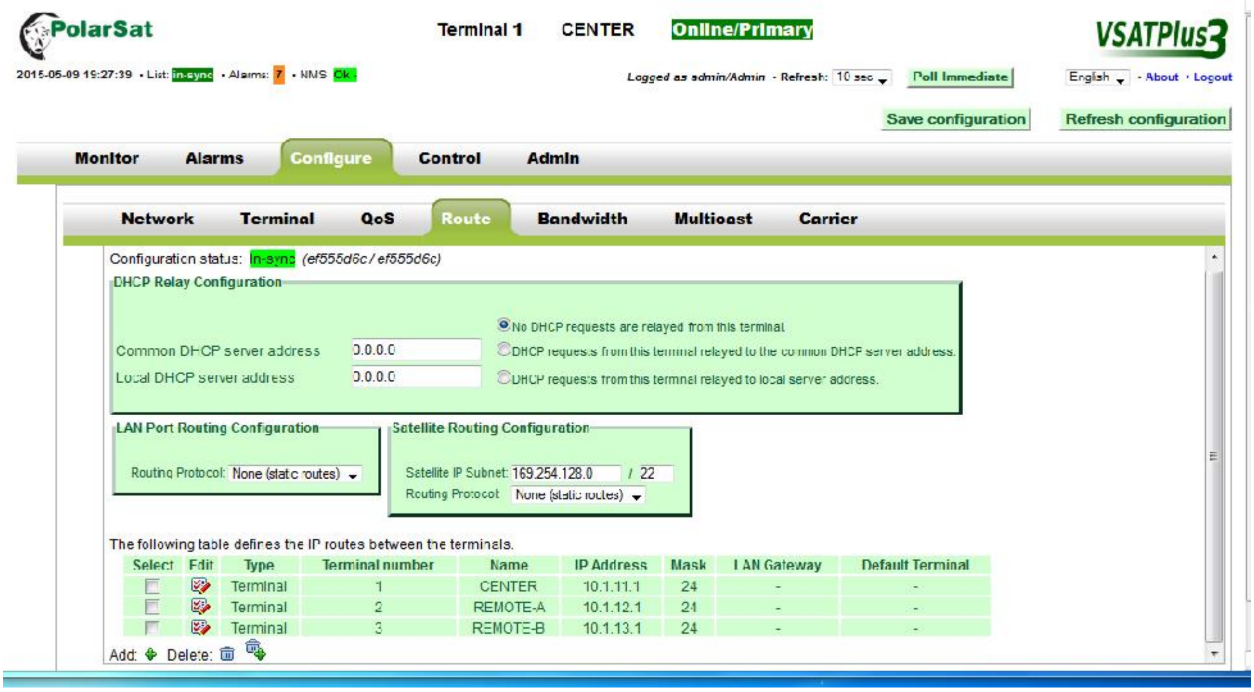


Figure 4-11: Network Statics IP Routing

Network TCP Configuration (only done once at primary station)

The figure below shows:

- Capture screen from CENTER (Primary Station) showing TCP Acceleration Setting

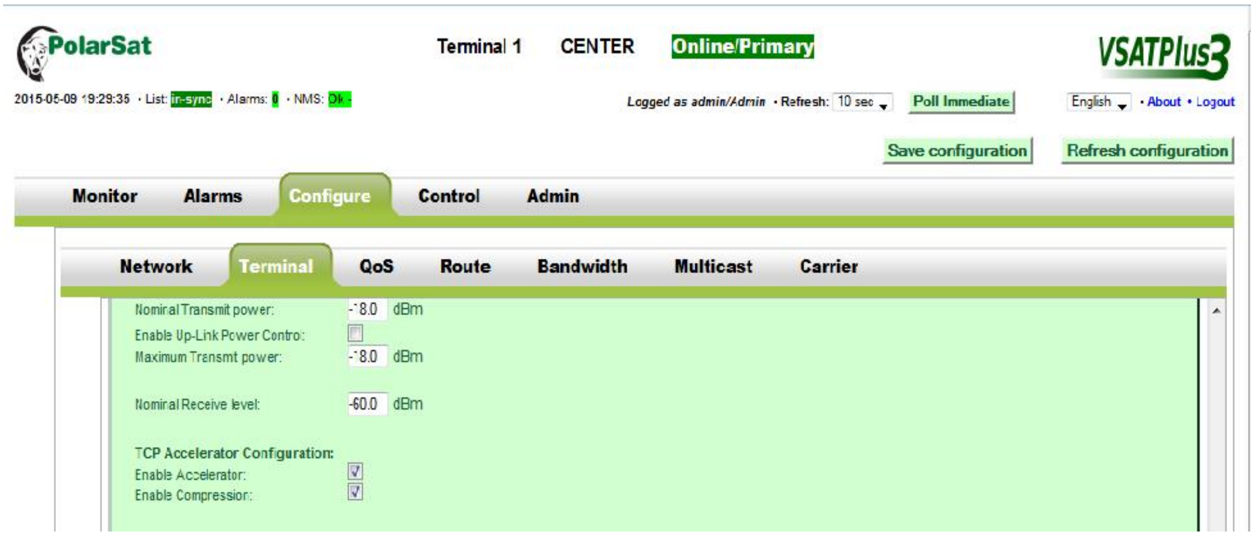


Figure 4-12: TCP Acceleration and Compression Configuration

B. REMOTE-A VSAT Station Terminal Parameters Status

The figure below shows:

- Capture screen from REMOTE-A (Secondary Station displaying its own Status showing that the station is ONLINE and IN SYNCH.

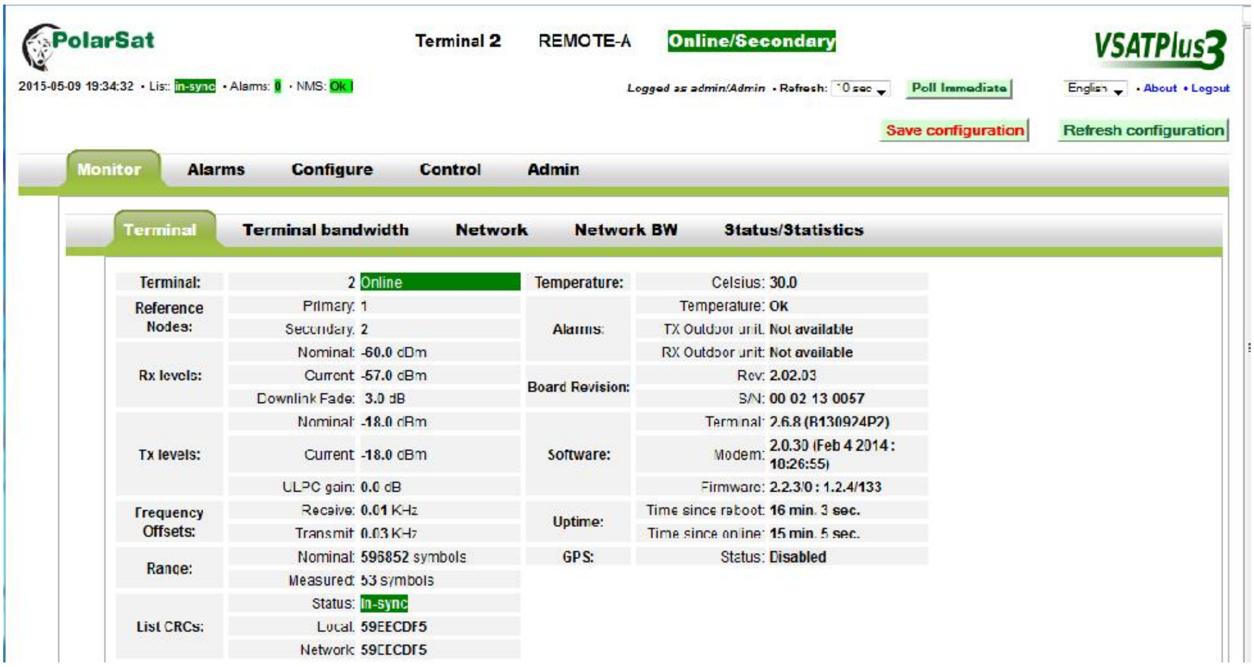


Figure 4-13: REMOTE-A Station Terminal Parameters Status

-

C. REMOTE-B VSAT Station Terminal Parameters Status

The figure below shows:

- Capture screen from REMOTE-B displaying its own Status showing that the station is ONLINE and IN SYNCH.

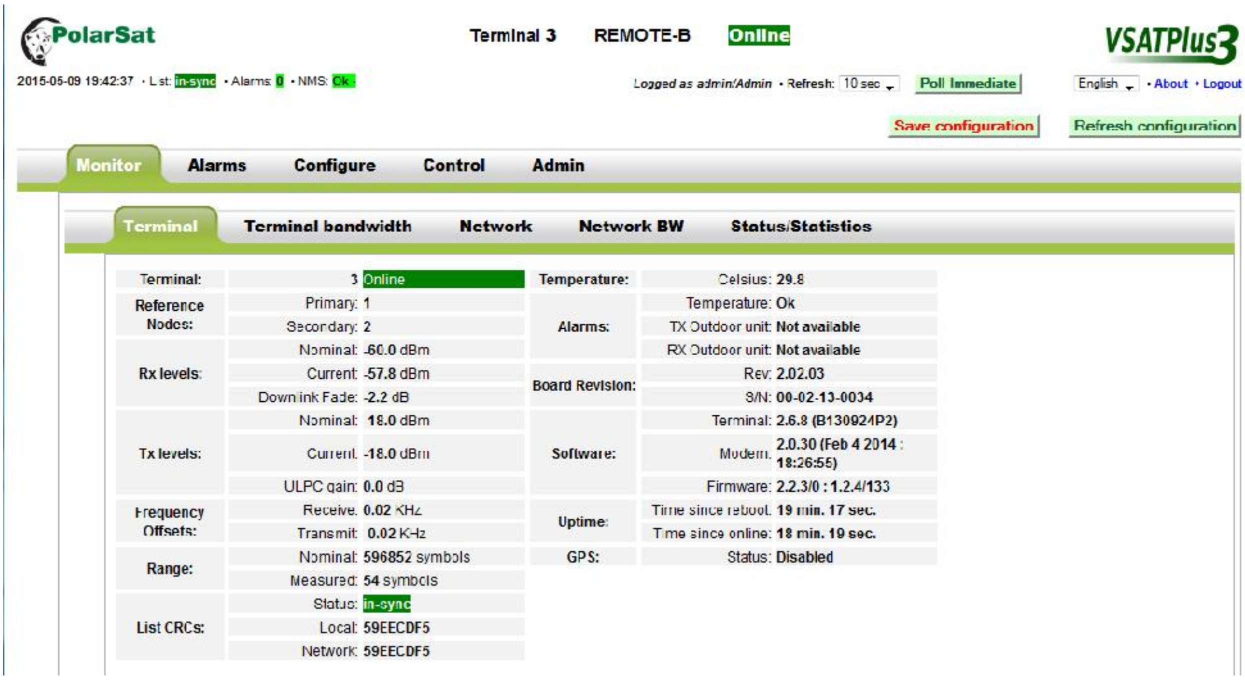


Figure 4-14: REMOTE-B Station Terminal Parameters Status

Multiplexers Network Over IP VSAT Network Connectivity

A. Multiplexers **LINK** Connectivity Over VSAT Network

The figure below shows:

- Capture screen from CENTER multiplexer displaying:
- Links Connectivity between CENTER Station and REMOTE Stations (on DATA mean the CENTER Site online with others station with a delay of 114ms and 13ms with the SERIAL)

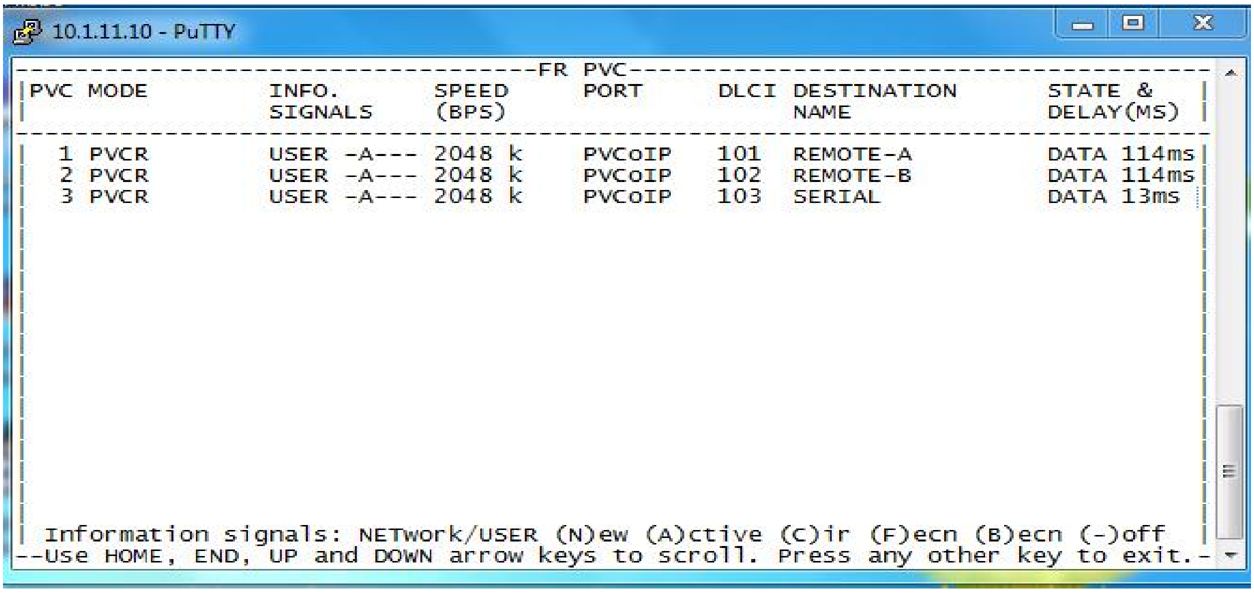


Figure 4-15: CENTER Multiplexer Link Connectivity to other Multiplexers Station

B. VOICE Ports Connectivity Over the VSAT Network

The figure below shows:

- Capture screen from CENTER multiplexer displaying:
- CENTER Site Voice Ports connectivity to REMOTE Sites ,IDLE (mean no Traffic but the port is configured , On Line (mean there are phones call for Center to Remote)

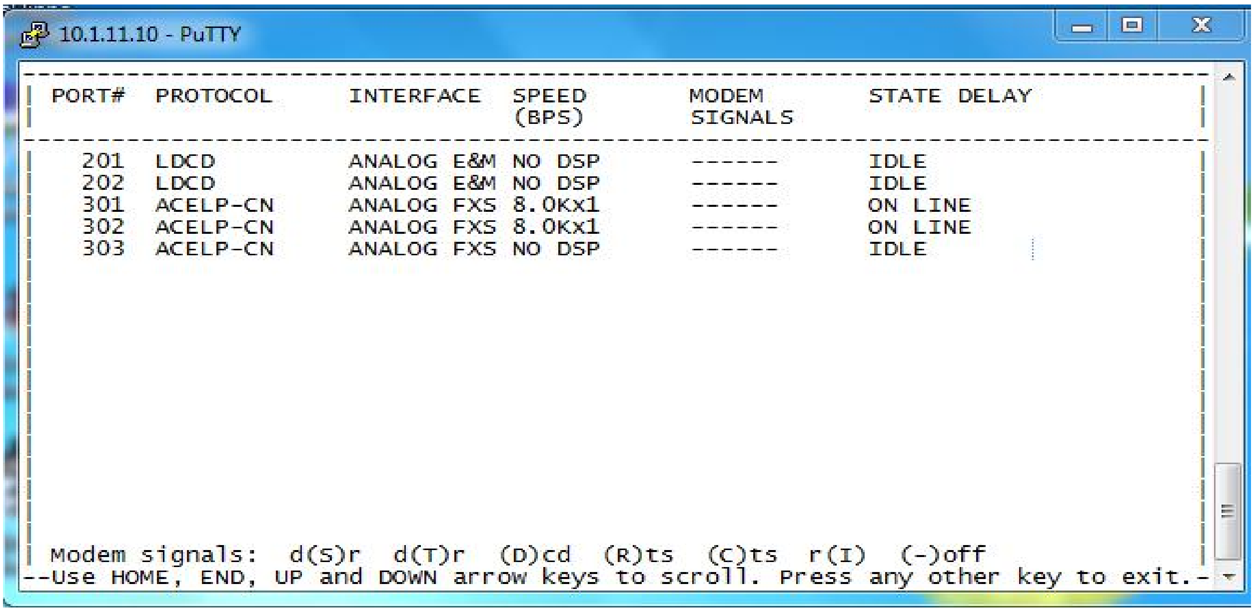


Figure 4-16: CENTER to REMOTES Station Voice(VHF + Telephones) Ports Connectivity

C. Multiplexers DATA Ports Connectivity Over VSAT Network

The figure below shows:

- Capture screen from CENTER multiplexer displaying two radars ports connection between the center site and the two remote sites (DATA means the port is online and ready for sending traffic over it)

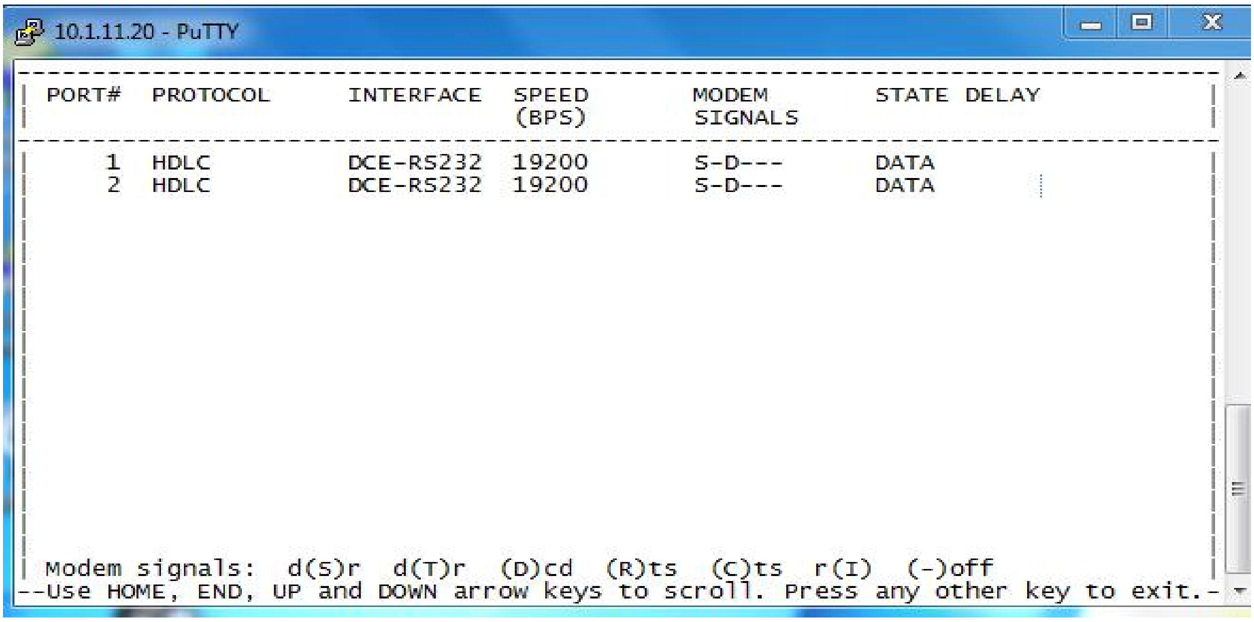


Figure 4-17: CENTER to REMOTES Data (Radars) Ports Connectivity

D. IP User Traffic Connectivity

The figure below shows:

- Capture screen from CENTER Multiplexer displaying IP Connectivity between **CENTER** Station and REMOTE-**A** and REMOTE-**B** Multiplexers Station for IP User Traffic.

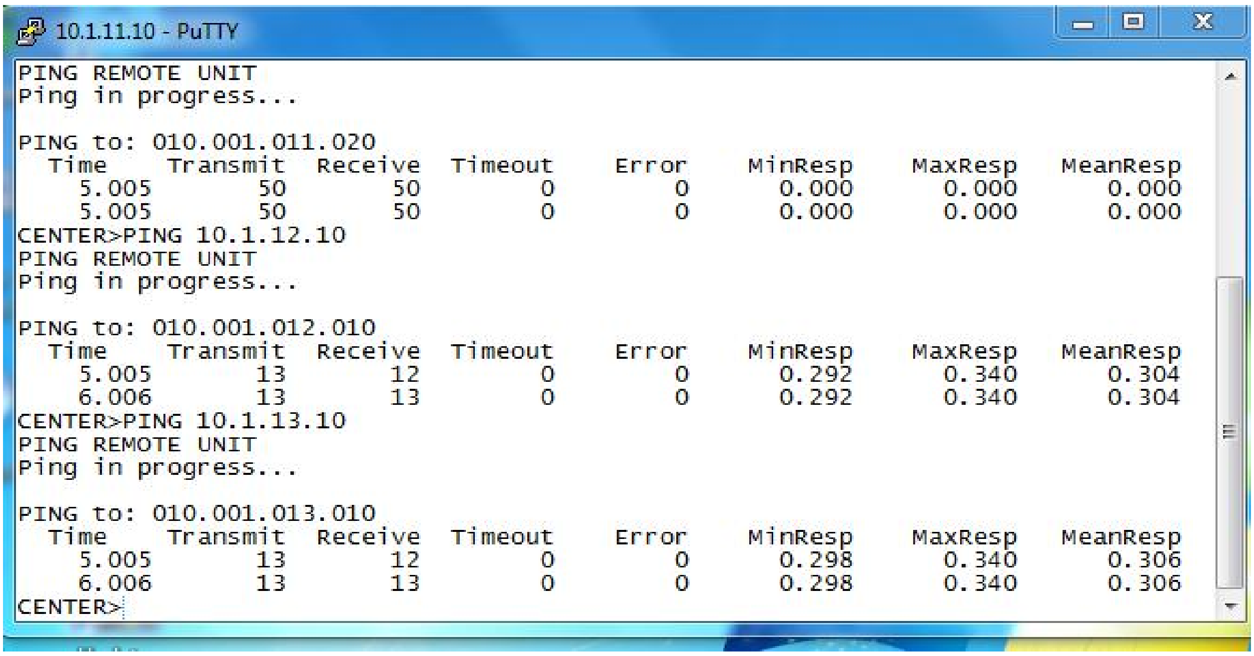


Figure 4-18: CENTER to Remote Stations IP User Traffic Connectivity

4.4 Results

4.4.1 Bandwidth Consumption

Voice Codec's Bandwidth Utilization

The codec's ACELP-CN (comfort noise) 8K is the codec of choice for Digital Circuit Multiplication Equipment (DCME) application.

Due to framing of the compressed voice, overhead must also be added to the overall voice rate. For example, ACELP8K-CN really takes 12.44Kbps of bandwidth while voice is active and transported on a leased line or Frame Relay links for example. And let us not forget the effect of comfort noise, reducing overall bandwidth utilization by around 40% when the silence suppression feature is activated.

Furthermore, one may configure the amount of buffering (single, double or triple) preferred for voice encapsulation. Double and triple buffering reduces bandwidth and Cell per Seconds (CPS) consumption per voice channel with an imperceptible change in voice quality.

1 CPS consumption is calculated by dividing one second by the voice protocol frame time multiplied by two (because of the full duplex nature of the voice conversation). Note that a fax transmission is half-duplex though.
2 Note that comfort noise reduces overall CPS utilization by around 40% when silence suppression is activated.
3 Using PVCR over IP with cell packetization enabled.

The following figures 4-19 and 4-20 summarize bandwidth requirements for each voice codec when transported over leased lines or IP networks.

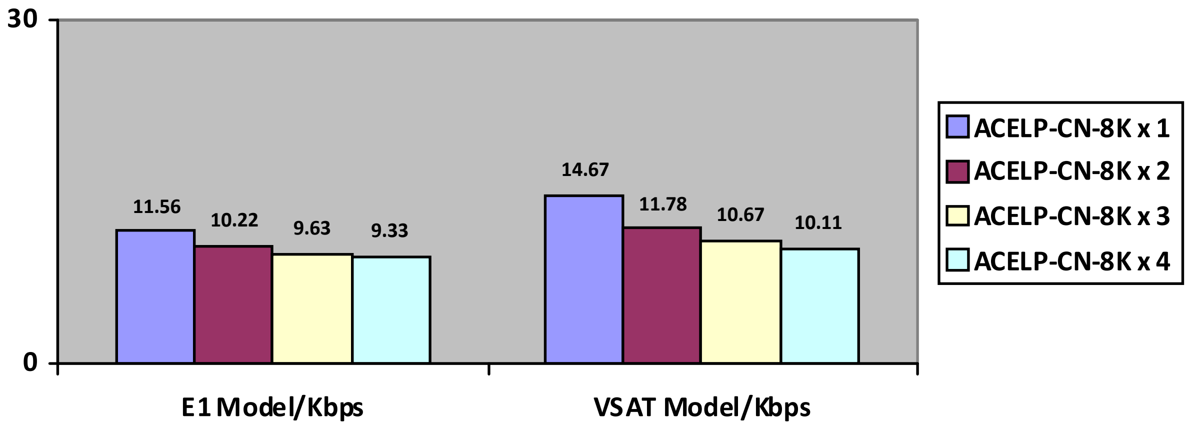


Figure 4-19 Voice (Telephony) Bandwidth Utilization (1)

Result (1-A) Practical implementation shows Voice Protocol/buffering ACELP8K×1 is the best for ATC environments showing good quality and not much bandwidth and it's suitable for sensitive voice applications (Hotlines).

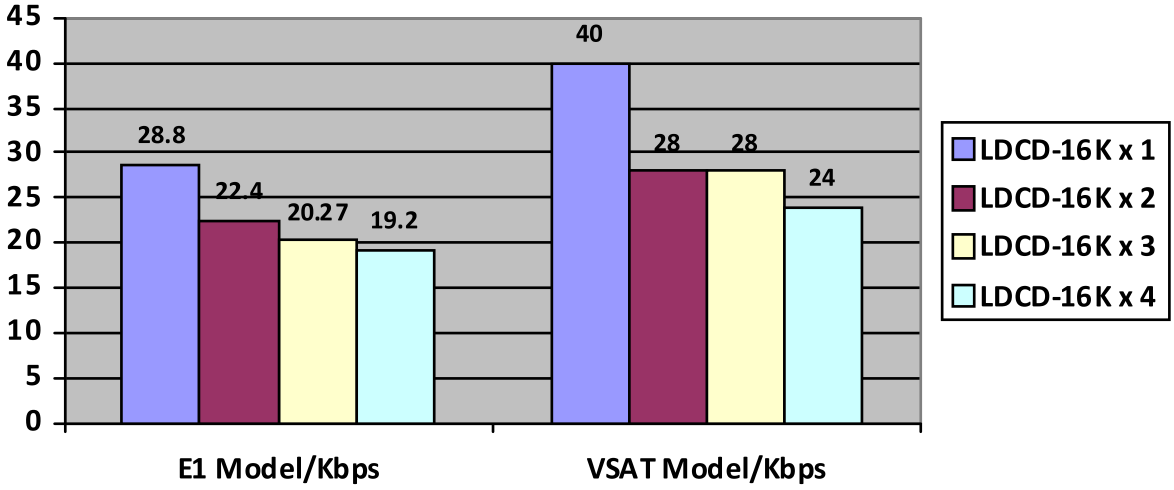


Figure 4-20 Voice (VHF) Bandwidth Utilization (2)

Result (1-B) Practical implementation shows Voice Protocol/buffering LDCD 16K×1 is the best for ATC environments showing good quality and not much bandwidth and it's suitable for sensitive voice applications (VHF Radio Services) .

4.4.2 Network Delay analysis and calculation

Table 4-1 Network Delay Analysis

	Delay
Input Buffer	18 ms [1]
Compression	1 ms
Access Queue	2 ms
Network Latency	25 to 250 ms [2]
Far End Queue	1 ms
Jitter Buffer	5 to 255 ms
Decoder	5 ms

1 Depending on Voice Protocol (refer to chart 5-3)
2 Network Latency may be 25 ms or higher for a Public Frame Relay Network and 250 ms for Satellite links

Models Network Link Delay

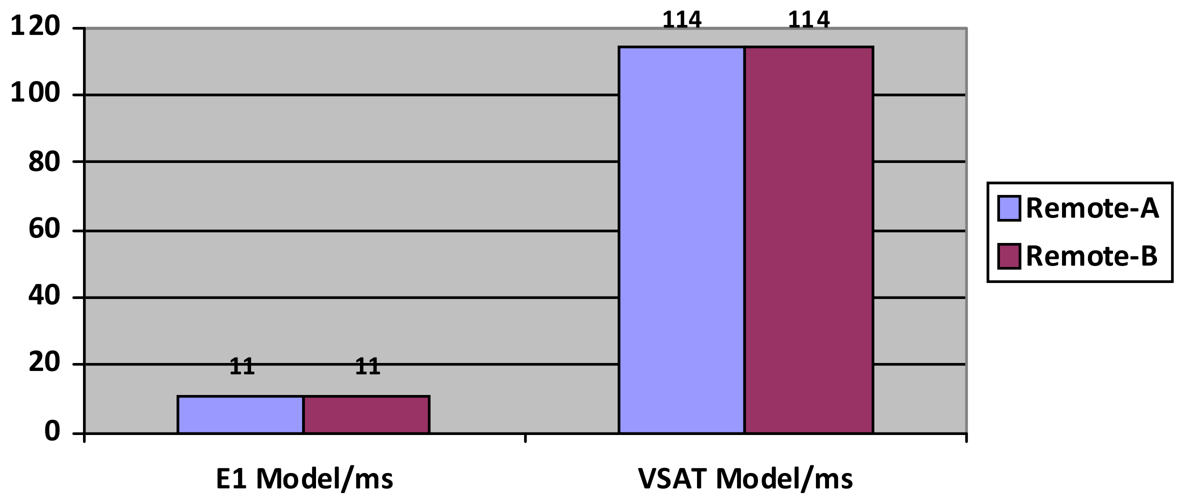


Figure 4-21 Models Network (Link) Delay Calculation

Result (2) Network Delay: Network jitter is defined as the variance in delay from one voice packet to the next. Several elements may contribute to jitter variances and delays, including switch latency within the network, queuing at the network entry and exit points, and voice digitization and compression. Following is the typical delay of a voice call using ACELP-CN 8k single buffering.

Network Delay: Data and IP traffic delay

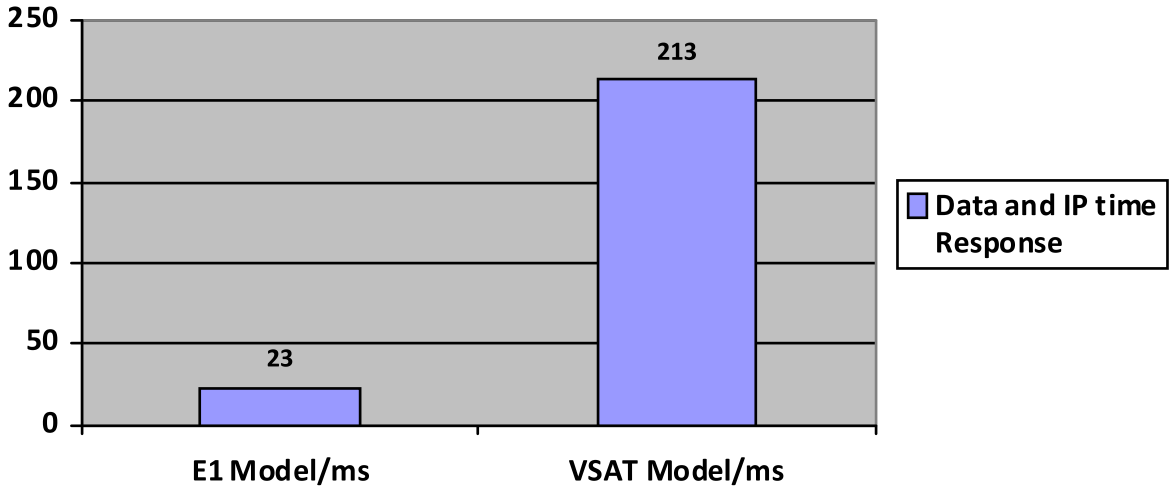


Figure 4-22 Models Network (Data) Delay Calculation

Result (3) Network Delay: Network jitter is defined as the variance in delay from one packet to the next. Several elements may contribute to jitter variances and delays, including switch latency within the network, queuing at the network entry and exit points, and Data compression. Above is the typical delay of a data and IP Traffic delay in the two models.

Chapter Five

Conclusion

In this chapter, we present the conclusion of the models Implementation results conclusions are provided for the performance of the models. Also the measuring of B.W utilization, capacity links, voice quality, and voice and data throughputs calculation is mentioned, and at the end of this chapter there is the future work.

5.1 Conclusion

This research presented a major challenge. Because the telecommunications network supported the exchange of critical voice and data traffic between aircraft (pilots), on-board aircraft navigation systems and the air traffic controllers and their consoles it needed to be available at all times. The system therefore had to provide very high levels of reliability for both equipment and services and needed to make use of diverse terrestrial (leased lines) and satellite communication technologies.

The implementation of the two models provides a complete system solution for organizations with wide area internetworking requirements. The Multiplexer maximizes bandwidth usage with high throughput levels, low overhead and minimal delays, and guarantees reliable integration of voice, fax/modem and data traffic. Multiplexer significantly cuts communications costs by cutting long-distance charges and the need for dedicated voice and data circuits in the network.

The two network models implemented voice and data routers combine the functionality of a data router, a voice gateway and a RAN backhauling optimizer in a single device, enabling the creation of digitally converged networks.

The two network models implemented can be used over public or private IP networks as well as Frame Relay, leased-line, Satellite or ATM. Well suited for large point-to-point and any-to-any networks, the Multiplexer uses a QoS cell based protocol called PVCR to efficiently integrate either voice/data or data-only solutions.

These networks are intended for a broad range of applications and serve the internetworking needs of central and remote sites. Individual products utilize a common hardware platform and are available as standalone products or scalable rackmount models with integrated data, voice and RAN backhauling (software option) capabilities over any networks.

There are several advantages of using the open IP standard instead of proprietary circuit-switched solutions. One major advantage is increased interoperability.

Using IP allows interconnecting commercial off-the-shelf components from various manufacturers. Special-purpose, expensive software and hardware can be replaced by off-the-shelf software and standard Internet equipment such as routers and switches. Using IP also enables integration with external data services such as weather information, and connections over a wide area network (WAN), so that controllers can perform their task from a remote location. Failure recovery is another important area where packet-switching has an advantage over circuit-switching. Failure of a node or link in a circuit-switched network would cause the communication to be interrupted, unless precautions had been made beforehand. On a packet-switched network, an alternative path for the packets will automatically be computed as a consequence of the normal operations of the network.

However, packet-switched networks also resolve all drawbacks caused by circuit switched infrastructures.

With the transition to broadband transmission infrastructures and the demand for interoperable radio-communications networks, air traffic control authorities are facing formidable tasks. The technologically superior VoIP-based communications systems can help manage these tasks.

Normally, complexity goes up and quality goes down as compression increases. ACELP-CN demonstrates that voice can be compressed as low as 6Kbps and still achieve near toll quality. With low-cost toll-quality voice compression algorithms and management of voice and data transmission parameters, voice quality can be maintained in high-traffic networks.

- **The Implementation of two models Shows that the Model one is the better than Model Two in compression with the Network delay , the VSAT has 250ms delay which is bigger than the delay in E1 Leased Lines over Fibber Optic (11-25 ms)**

5.2 Future Work

Voice over IP (VoIP) is the technology of the future in air traffic control (ATC) and air defence (AD) sectors. After years of transferring radar data for air traffic management over IP networks, IP technology is now expanding its way into voice communications. Reduced infrastructure costs are one benefit: If a common network is used to transport voice and data, only a single network needs to be planned, installed and operated. The trend towards VoIP will be accelerated by the successful standardization of VoIP in the ATC sector by the adoption of these standards by the ICAO.

In other words, VoIP technology development and VoIP VHF will have to prove in the near future that they are ready to meet the requirements of more flexible, safer and cheaper way of air traffic management.

In the new fully IP-based voice and data communications network. ATC organizations can now rely on voice communications systems and air traffic data from a single source – from the managements systems to the antennas.

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