Fiber optics lab

1. Start the Simtel 2.0 software
2. Choose the Electromagnetic spectrum option from the Basic Laws menu
    a. Start the animation
    b. Follow the frequency ranges and note the range limits, the energy of the signals and their applications
3. Choose the Snell's Law option from the Basic Laws menu
    a. Select successively the possible variants for the two environments, the light source, and change the angle of incidence. The resulting parameters for each individual simulation should be noted
4. Choose the Acceptance Angle option from the Basic Laws menu
    a. Change the angle of incidence and note the interval for which the total reflection is obtained
5. Choose the Multimode Graded Index Fiber option from the Types of Fiber menu
    a. Change the angle of incidence and observe the propagation through the fiber
6. From the Attenuation losses menu, choose the option Dispersion -> Chromatic Material Dispersion
    a. Change the wavelength of the signal and note its influence
7. Choose the Optical Sources -> LED and Laser option from the Parts of Fiber Optic Communication System menu
    a. Note the influence of different parameters on the signal
8. From the Parts of Fiber Optic Communication System menu, choose the option Diodes -> Avalanche Photodiode
    a. Note the influence of the different parameters on the photodiode’s operation mode
9. Choose the -> Fiber Optic Communication option from the Parts of Fiber Optic Communication System menu
    a. Select the input sequence (bits)
    b. Change the type of light source
    c. Note the transmission of information
10. Choose the -> Wavelength Division Multiplexing option from the Fiber Optic Communication menu
    a. Select the input sequence (bits) for channel 1
    b. Select the signal wavelength for channel 1
    c. Select the input sequence (bits) for channel 2
    d. Select the signal wavelength for channel 2
    e. Select the input sequence (bits) for channel 3
    f. Select the signal wavelength for channel 3
    g. Start the simulation for different configurations and write down the results obtained.

GMDSS system

History and evolution of GMDSS

In 1899, radio communications were used for the first time to save lives in danger at sea. Since then, various organizations have developed this way of communication to protect human lives at sea. These bodies always use the best tools technology has to offer , at all times.

Since February 1, 1992, the use of new technologies has produced a profound transformation of the system used. Since then, GMDSS has been in operation, involving the automation and widespread use of satellites (the INMARSAT network - International Mobile and Maritime Satellite Organization) for communications.

The system is supported by the International Maritime Organization, with work beginning in the 1970s. In addition to communications through the INMARSAT system, other essential services were created, such as the navigation alarm service, auxiliary services such as NAVTEX, the standard vocabulary of navigation and search and rescue services from many countries.

In 1983, and later in 1987, amendments were adopted to regulate the radio communications of the ITU (Union the International of Telecommunications), which established frequencies, procedures, types of broadcasts and personnel for radio , including GMDSS in the aid procedures. After several subsequent amendments, the full implementation of the system was achieved in 2005.

GMDSS System Overview

The GMDSS system is based on the use of the most modern technologies in radio telecommunications to achieve an integrated global system of communications in case of danger, related to the safety of navigation, between ships or ship-to-shore, anywhere in the world.

In GMDSS, the basic concept is that the land authorities responsible for search and rescue, as well as other vessels in the vicinity of the one in danger, will be able to take part in SAR operations in the shortest possible time.

The system also provides emergency communications, safety of navigation, weather and navigation warnings, weather forecasts or other urgent information for ships. Each ship, depending on the area in which it operates, must be able to perform those communication functions that ensure the safety of the ship and the safety of other ships sailing in the same area.

The GMDSS system is an automated system. Alerts can be transmitted in three directions: ship-to-coast, ship-to-ship and ship-to-coast, in all marine areas. Thus, an attempt is made to increase the efficiency of search and rescue in emergency situations, to improve information and act more quickly.

The GMDSS system uses INMARSAT geostationary satellites, SHOTS polar orbiting LEO satellites and Russia's COSMOS system.

The GMDSS system works at:

- all cargo ships of more than 300 tons;
- all passenger ships on international voyages;
- all fishing vessels built after 1 February 1999 and over 24 meters in length;
- all fishing vessels that are over 45 meters in length.

GMDSS components

The main types of equipment used in GMDSS are:

Emergency position-indicating radio beacon – EPIRB (Emergency Position Indicating Radio Beacon)

COSPAS-SARSAT is an international search and rescue system using satellites , established in Canada, France , the United States and Russia. These four countries helped develop the EPIRB on 406 MHz, an element of the GMDSS system intended to work with the COSPAS-SARSAT system. The self-activating EPIRB system, currently mandatory on SOLAS vessels, fishing vessels and all passenger vessels, is designed to transmit alerts to rescue centers via satellites. The original COSPAS-SARSAT system used orbital satellites, but in recent years the system has been expanded to include 4 geostationary satellites . Systems that include GPS receivers transmit extremely precise positioning information (with a deviation of about 20 meters).

NAVTEX

international , automated system for the instant distribution of Maritime Safety Information (MSI) to ships , including navigation warnings, weather forecasts and warnings, search and rescue notifications and other similar information . The system contains a radio receiver on board the ships that checks each received message to determine its importance to the ship's master. The transmission frequency of these messages is 518 kHz for English, while the 490 kHz frequency is sometimes used for broadcasting in a local language. The messages are coded and contain a header that uses letters of the alphabet to represent the broadcasting stations, the type of messages . These are followed by two digits indicating the sequence number of the message. For example, for the **FA56 header** , ***F is the*** transmitting station ID, A indicates ***a*** warning message, and ***56*** is the message sequence number.

Enmarsed

The satellite systems operated by Inmarsat , supervised by IMSO (International Mobile Satellite Organization) are also important elements of GMDSS. The types of Inmarsat terminal stations recognized by GMDSS are: Inmarsat B, C and F77. Inmarsat B and F77, an upgraded version of Inmarsat A, provide ship-to-shore, shore-to-ship or ship-to-ship communications. Telephone, telex and high-speed data services are available, including priority distress alerts, telephone and telex, at- and from rescue coordination centers.

Inmarsat C provides ship-to-shore, shore-to-ship or ship-to-ship communications via e-mail messaging, has the ability to transmit pre-formatted alert messages to a rescue coordination center and offers the Inmarsat C SafetyNET service . The Inmarsat C SafetyNET service transmits weather, navigation , radionavigation warnings , ice reports and other similar information not provided by NAVTEX. SafetyNET works similarly to NAVTEX, in areas outside its coverage area .

SOLAS currently requires INMARSAT C equipment to have an integrated receiver for navigation satellites, or to be connected to an external receiver. This requirement will ensure the transmission of exact location information to the rescue coordination center if an alert signal is issued.

High Frequency

A GMDSS system may include radiotelephone and high frequency (HF) radiotelex (narrowband, direct printing). Calls are initiated by a digital device for selective calling (DSC - Digital Selective Calling).

Locating device for S&R operations

The GMDSS installation on ships includes one (two on ships over 500 tonnes) search and rescue locating device called Search and Rescue Radar Transponder (SART and Rescue Radar Transponders), which is used to locate the ship by creating a series of twelve dots on the rescue ship's 3 cm radar screen. The detection distance is normally about 15 km (8 nautical miles). Once detected by the radar, the SART will generate a visual and acoustic indication for persons in distress.

Digital Selective Calling

Digital Selective Calling (DSC), using MF, HF and VHF, is part of the GMDSS system. DSC has the primary role of initiating telephone calls and MF/HF radiotelex for ship-to-ship, ship-to-shore and shore-to-ship communications. DSC calls can be made to individual stations, groups of stations, or "all radio stations" in range. Each DSC equipment is assigned a unique 9-digit number (MMSI - Maritime Mobile Service Identity).

DSC distress alerts, which consist of a pre-formatted message, are used to initiate emergency communications with ships and rescue coordination centres. DSC was designed to eliminate the need for a person to continuously monitor radio receivers on voice channels, including channels 16 VHF (156.8 MHz) and 2182 kHz - now used for distress, security and calling.

Functional requirements of a ship with GMDSS capability

9 main communication functions are defined for ships that are at sea:

1. Transmit ship-to-coast distress alert by at least 2 separate and independent means, each using a different radio communication service
2. To receive coast-to-ship distress alert
3. Transmit and receive ship-to-ship distress alerts
4. To transmit and receive information for the coordination of search and rescue operations
5. To transmit and receive communications at the wreck site
6. To transmit and receive signals for the purpose of localization
7. To transmit and receive information regarding maritime security (MSI – Marine Security Information)
8. To transmit and receive general communications intended for terrestrial radioelectric networks or systems
9. To transmit and receive general information between ships.

Allocated frequencies in the maritime mobile service and the maritime mobile satellite services

The frequencies allocated in the maritime mobile service are MF, HF, VHF, and for the satellite service they are L, C, respectively from UHF and SHF. Allocation of frequencies in the maritime mobile service:

Frequency bands	**Services**
415÷535 kHz	Morse radiotelegraphy, emergency call frequency: 500 kHz
518 kHz – for English	NAVTEX frequency - integral part of DMDSS, for the transmission of navigation and weather warnings, urgent navigation safety messages
490 kHz	NAVTEX for national languages
1605÷4000 kHz	Telephone, TELEX and NBDP services
2182 kHz	Call frequency in case of alert in RT
2187.5 kHz	Call frequency in case of DSC alert
4 MHz÷27.5 kHz	RT HF services; public correspondence service; DSC transmissions in case of alert; weather and navigation warnings; communications between ships
121.5 MHz	EPIRB in civil aeronautical service; can be used in SAR operations by aircraft flying over the event area
156÷163 MHz	RT VHF service; ship traffic control services; port services; communications between ships; transmission of local weather and navigation warnings
Channel 6 in VHF - 156.3 MHz	Transmissions from ship; ship station - aero radio station communications in the coordination of SAR operations; radio traffic for navigation safety; ship station - aero radio station communications ; communications between two air stations in SAR operations
Channel 13 in VHF - 156.650 MHz	Ship-to-ship communications regarding navigational safety; ship-shore communications
Channel 70 (DSC Channel) in VHF - 156.525 MHz	Transmission of alerts in case of danger; radio traffic for navigation safety; radio traffic in air navigation for navigation safety communications
Channel 16 in VHF RT - 156.8 MHz	International channel for radio traffic in RT; communications in case of alert; emergency call in RT authorized by the maritime mobile service in VHF
Channel 26 in VHF: 161.9 MHz for coast stations 157.3 MHz for ship stations	General ship-to-ship, ship-to-coast communications via coast stations; commercial information transmissions
243 MHz	EPIRB call in aeronautical service, military applications
216÷220 MHz	Extended telephone services, public correspondence, US maritime telephone services
406÷406.1 MHz	COSPAS-SARSAT EPIRB
457.525÷467.825 MHz	Shipboard Communications (UHF)
1530÷1559 MHz 1625.5÷1660.5 MHz	Frequency bands allocated to the maritime mobile satellite service
9.3÷9.8 GHz and 3 GHz	Radar and transponder radar search and rescue

Maritime areas

The GMDSS divides the seas and oceans into 4 maritime zones. These areas determine the equipment that a ship must have, as well as the specialization of the personnel who will operate this equipment.

The maritime areas are:

- A1: Area covered by VHF coast stations, where alerts are monitored continuously; extends up to 25-30 Mm;
- A2: Area covered by MF coast stations, with continuous alerting service; it stretches between 25 and 100 Mm from the shore;
- A3: Area covered by INMARSAT satellites, with continuous alerting service; covers the area 70°-75°N and S;
- A4: Polar areas, as well as any other area outside of areas A1, A2 and A3

The equipment of the ships according to the navigation areas

Zone A1:

- VHF radio telephone + DSC controller dedicated to RT VHF communications and transmission/reception of DSC alerts;
- DSC watch receiver on channel 70 dedicated to receiving alerts transmitted on VHF channel 70;
- RT 2182 kHz watch receiver for receiving alerts transmitted in RT MF;
- NAVTEX receiver for receiving MSI information;
- COSPAS-SARSAT or INMARSAT EPIRB for location in case of alert.

Lifeboat equipment:

- portable VHF radiotelephone for RT communications at the place of the alert;
- radar transponder for location signal.

Area A1+A2:

- VHF radio telephone + DSC controller dedicated to RT VHF communications and transmission/reception of DSC alerts;
- DSC watch receiver on channel 70 dedicated to receiving alerts transmitted on VHF channel 70;
- RT 2182 kHz watch receiver for receiving alerts transmitted in RT MF;
- NAVTEX receiver for receiving MSI information;
- EPIRB COSPAS-SARSAT or INMARSAT for location in case of alert;
- MF/HF radio system

Lifeboat equipment:

- portable VHF radiotelephone for RT communications at the place of the alert;
- radar transponder for location signal.

Area A1+A2+A3 (variant with INMARSAT):

- VHF radio telephone + DSC controller dedicated to RT VHF communications and transmission/reception of DSC alerts;
- DSC watch receiver on channel 70 dedicated to receiving alerts transmitted on VHF channel 70;

- RT 2182 kHz watch receiver for receiving alerts transmitted in RT MF;
- NAVTEX receiver for receiving MSI information;
- EPIRB COSPAS-SARSAT or INMARSAT for location in case of alert;
- MF/HF radio system;
- INMARSAT-A ship station with separate or embedded EGC receiver which is additional, intended for communications of any type (telex, telephony, e-mail, data transmissions) or INMARSAT-B coast station (digital variant of INMARSAT-A) . The EGC receiver is designed to receive MSI information.
- The SES INMARSAT-C station is intended for alerting by any means, except voice telephony.

An alternative for the A1+A2+A3 zone is the provision of a HF radio station, which is also valid for the A1+A2+A3+A4 zone, having in addition a radiotelex module (TOR) for telex communications and telex alerting.

Area A1+A2+A3 (variant with HF) and/or area A1+A2+A3+A4:
- VHF radio system (receiver and transmitter) + DSC controller;
- watch receiver on channel 70 VHF DSC;
- MF/HF radio systems + DSC MF/HF controller + radiotelex ;
- MF/HF DSC watch receiver designed to receive alerts in these bands;
- watch receiver on 2182 kHz;
- EGC mode (antenna without ship station);
- NAVTEX;
- COSPAS-SARSAT EPIRB or EPIRB-E.

Maris software

Simulation of marine radio communications

In order to simulate the real conditions of marine radio communications, several factors affecting the transmission and reception of radio signals were taken into consideration:

- transmission noises,
- different forms of radio wave propagation (direct visibility, terrestrial, ionospheric),
- the sound effects of different communication equipment,
- frequency of bandpass filters,
- frequency shifts,
- changes in reception, as a result of changes in the distances between radio stations,
- communications affected by day and night,
- the maximum and minimum frequencies, with which a communication can be carried out, taking into account the position of the transmitting and receiving stations, the solar activity, the day of the year and the time of transmission,
- multiple passes of the radio wave, as a result of the ionospheric reflection (Doppler effect).

Simulation of GMDSS equipment in Vox Maris

The simulated equipment is as follows:

- VHF radio

- DSC VHF
- HF radio
- DSC HF
- NAVTEX receiver
- Inmarsat C
- Inmarsat B
- radio telex
- EPIRB
- slap
- radar
- WatchReceiver 2182

With the aforementioned equipment, the following operations can be performed:

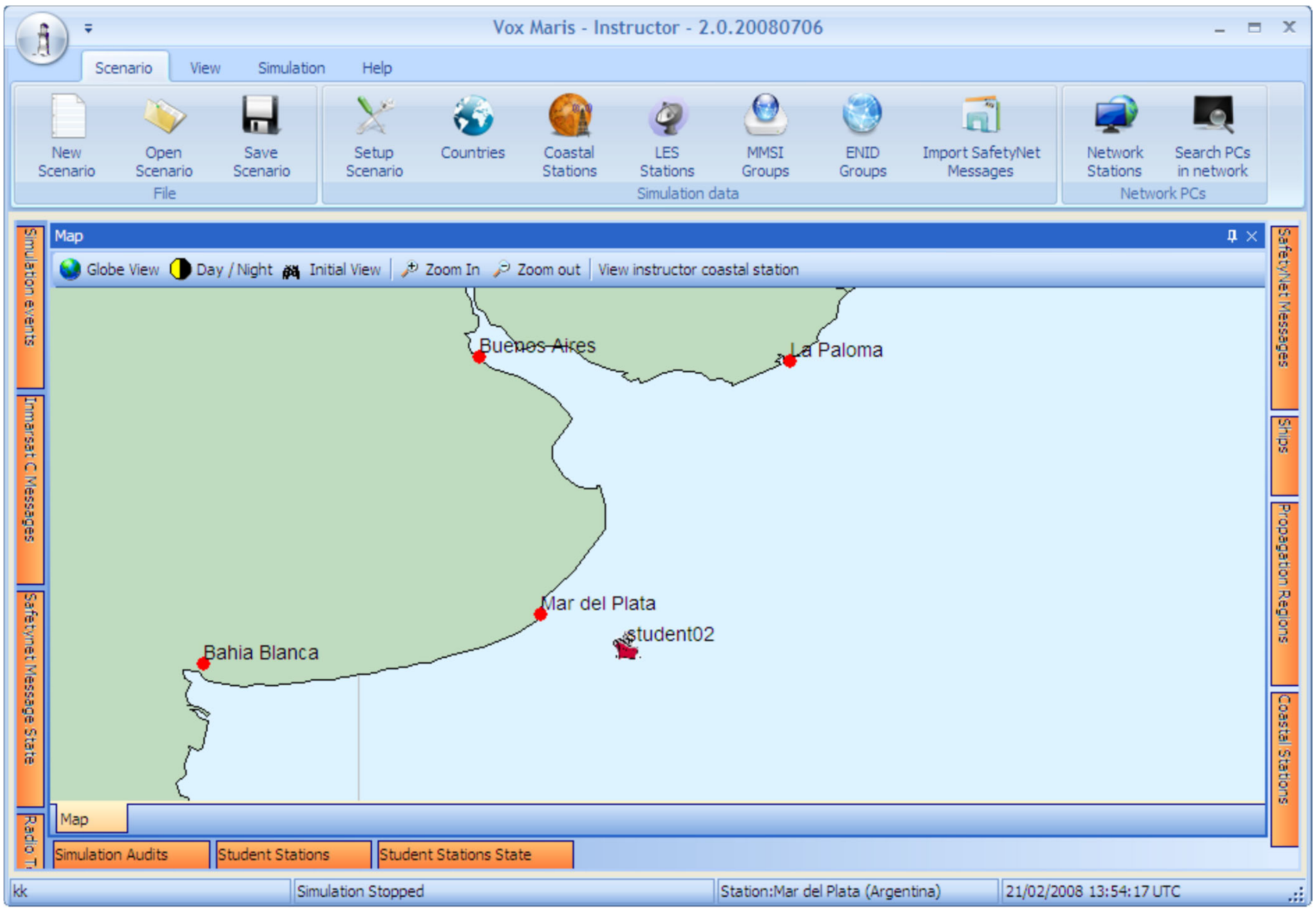
- the transmission of ship-to-coast alerts by means of at least 2 separate and independent channels, using different radio communication services,
- receiving ship-to-ship and coast-to-ship alerts,
- establishing communications for coordinating search and rescue operations
- location of ships
- coordination of operations in the wreckage area
- transmission and reception of maritime security information (MSI)
- exchange of information between ships and between ships and coast stations

Basic concepts

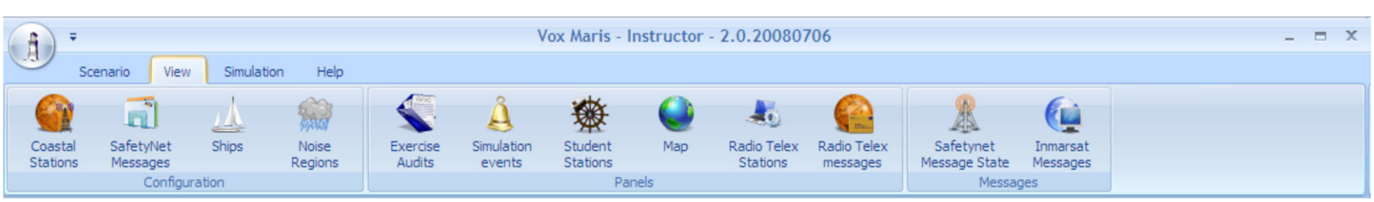
Instructor interface

Vox Maris Instructor has a series of mobile panels that contain different functionalities. Their position on the screen can be changed and this is saved for subsequent starts. The main menu of the application is separated into groups, with the following categories available:

- **Schemario** (***Scenario***): Options for configuring the scenario
- **View** : To ***display*** a panel that was accidentally closed
- **Simulation** : To control the simulation **.** In this category are options to start or stop the simulation or to send messages for Inmarsat C
- **Help : In this** category are user manuals and manuals for simulated devices **.**



If a panel was accidentally closed, select which panels will be displayed from the " View " group.



Access to simulated equipment for coast stations

When starting the simulation, the equipment will be displayed according to their availability for the selected coast station.

There are options to switch to the screens corresponding to the coast station equipment.

To return to the main screen, press the "Show Scenario " button.

At the bottom is the status bar, where the name of the scenario, the status of the simulation, the selected coast station, and the current date and time (UTC) are displayed.

Procedure:

1. The information available for the coast stations is noted
2. The information available for the vessel is noted
3. The equipment available for communications is noted
4. The mode of operation of the communication equipment - especially the mode of transmission of alerts - is regulated
5. Each group of students (7 groups) chooses a piece of equipment and prepares - according to its manual - a summary of how it works.

Voice over Internet Protocol

1. VoIP Analytical Simulator – short description & case study

The “VoIP Analytical Simulator” software aims to facilitate the implementation of data transfer and delay analysis algorithms for any network of interest. The simulator interface uses drag-and-drop workflow to build any network topology. The simulator includes functions related to files, editing, viewing and drawing. Figure 1 shows the main user interface of the VoIP simulator. It has two windows: the left panel and the right panel.

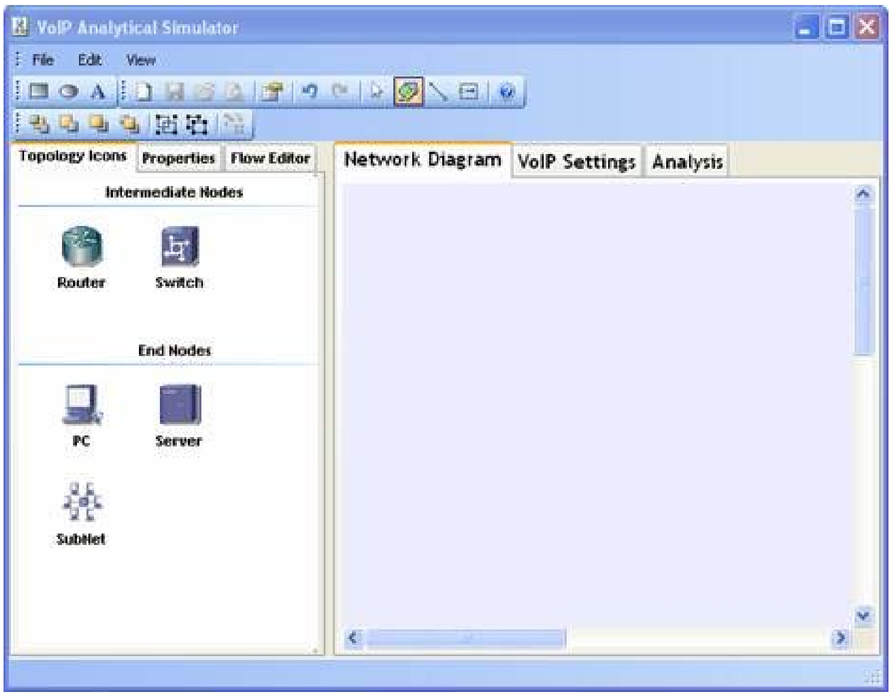


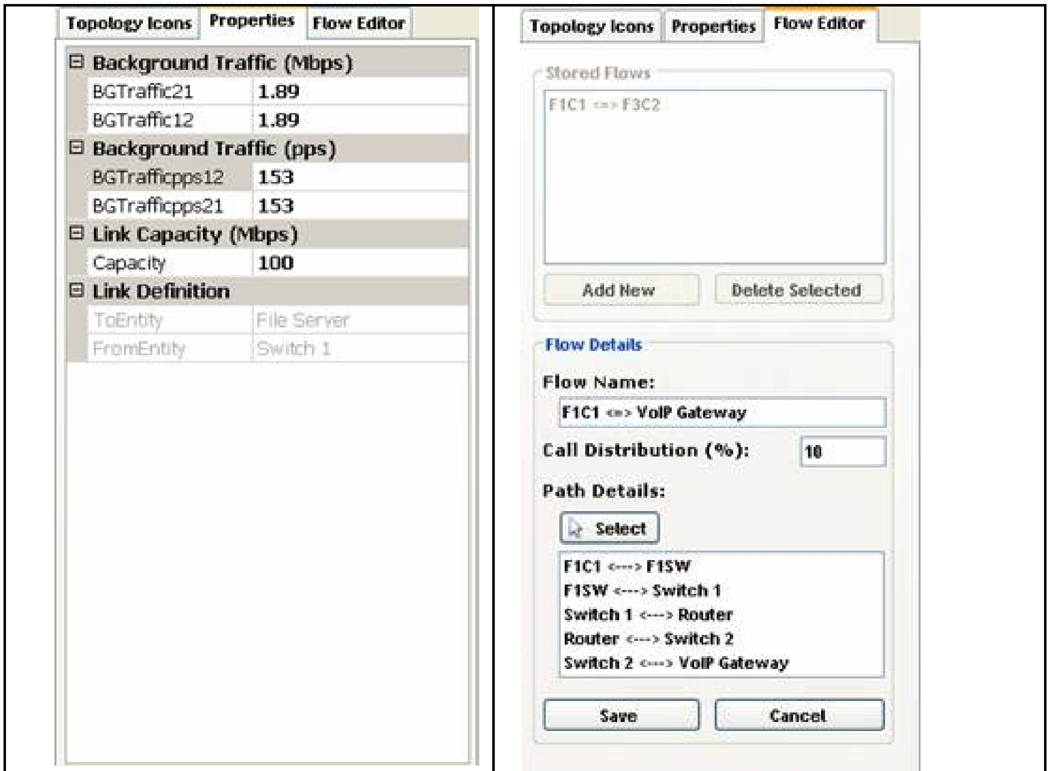
Figure 1. VoIP main interface

Left panel

The left panel has three tabs: *Topology Icons*, *Properties* and *Flow Editor*. Figure 1 shows the Topology Icons, where you can find all the elements used to build the desired network topology. Routers, switches, PCs, servers or subnets can be used. The network topology is obtained by dragging the desired elements into the panel on the right. After placing network elements, they can be connected by selecting the connection line icon on the bar at the top of the main simulator screen.

Figure 2(a) shows the *Properties* tab, which is used to specify the properties of network elements like nodes, links, and subnets. You can specify name, bandwidth or capacity, location, width, height and traffic. The properties of different network elements can be selected by clicking on the respective elements in the network topology. Each network element has different properties with default values. The sizes and placement of network element icons can be changed by changing their width and height in the properties window. Figure 2(a) shows the properties of an element that displays the capacity but also the traffic in Mbps and pps (packets per second).

Figure 2 (b) shows the *Flow Editor*, which is used to define VoIP flows or network call paths. A new stream can be added by clicking on the "*Add New*" button and then entering a name for the stream. Next, to specify the flow path, press "*Select*". The details of the route are simply entered step by step, in the network topology already drawn in the right panel, by clicking successively on the elements in the network. The constructed flow is bidirectional. The percentage of total calls that pass through this path is specified under the heading "*Call Distribution (%)*". To validate the created flow, press "*Save*". For example, Figure 2(b) shows the path details and percentage of a flow named "*F1C1-VoIP Gateway*". After pressing the "*Save*" button, this path will appear in the "*Stored Flows*" list. The call distribution of all streams must not exceed 100%.

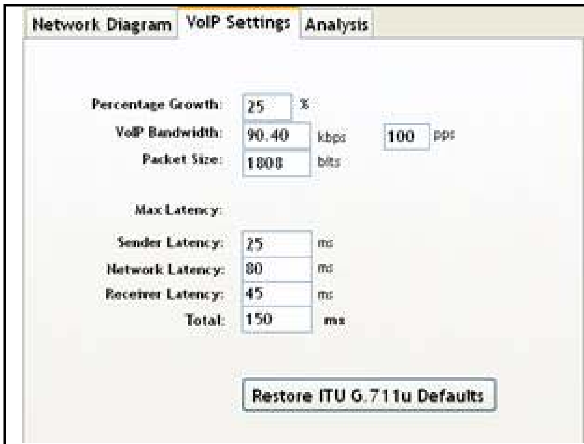


(a) (b)

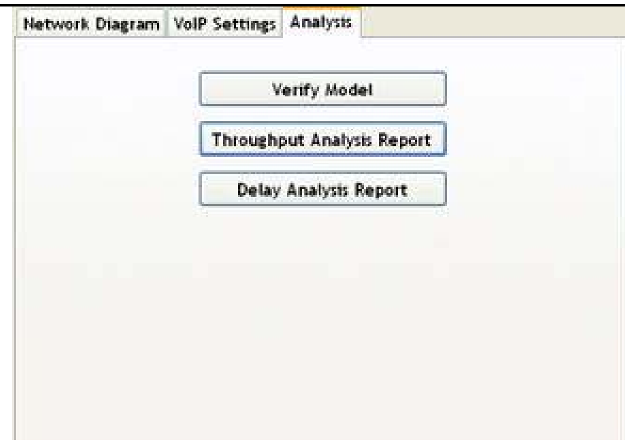
Figure 2. Properties of network elements (a) and VoIP flow editor (b)

Right panel

The right panel has three components, namely *Network Diagram*, *VoIP Settings* and *Analysis*. Figure 1 shows the *Network Diagram* panel, which is used to build the network topology. This is developed according to the previously described method. Figure 3(a) shows the default VoIP settings according to the ITU G.711u standards, specifically for bandwidth, packet size and latency. Users can restore or modify these settings. The "*VoIP Settings*" window allows users to set the future growth percentage of each network element, as well as the separation of the total 150ms delay into three delays, associated to sender, receiver and network. The results can be obtained through the "*Analysis*" tab shown in figure 3(b). Before obtaining the results, the user can inspect the model by pressing "*Verify Model*" button, to check the settings and configuration for errors. Errors can include missing links in the network topology or missing configurations. There are two different analytics reports that show the number of calls that can be supported. One ratio is based on transfer analysis and the other is based on delay analysis. Details of these reports will be presented below.



(a)



(b)

Figure 3. VoIP settings (a) and checks and reports (b)

The following example is intended to illustrate the simulator options. Figure 4 shows a typical network topology for a small business located in a multi-story building. The enterprise has a network on three floors. The network shows the VoIP nodes of a *gatekeeper* and H.323 *gateway*. The *gatekeeper* node handles signaling for establishing, terminating, and authorizing voice call connections. The VoIP *gateway* is responsible for converting VoIP calls to/from the PSTN (Public Switched Telephone Network). Other hardware requirements include a VoIP client terminal, which can be a separate VoIP device (eg IP phones) or a PC or workstation that has VoIP capability. The capacity of the router and switch is 25,000 pps and 1.3M pps, respectively. All links are full-duplex, Fast Ethernet (100Mbps).

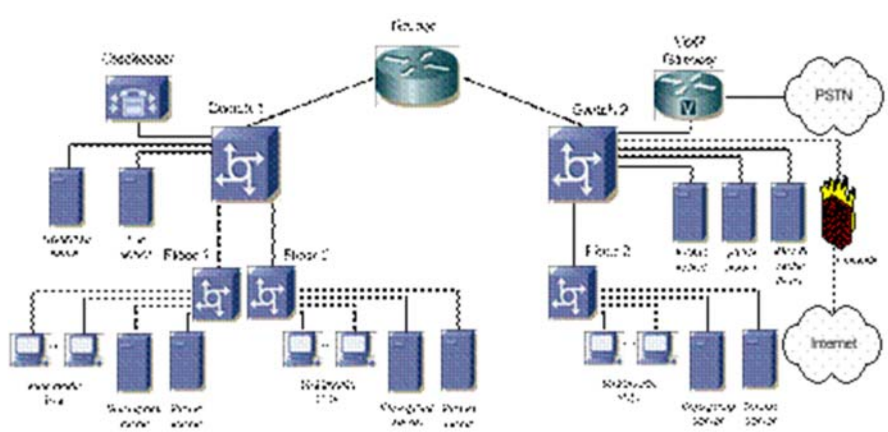


Figure 4. Network topology with VoIP components

Figure 5 shows the network model built in the simulation program corresponding to the network topology in figure 4. To avoid introducing numerous PC nodes or IP phones per floor, representing the end users (and therefore cluttering the network topology diagram), LANs of on the floors were modeled as a single LAN comprising an Ethernet switch and three PCs, used to model user activities. For example, Floor 1 has three nodes (labeled as F1_C1, F1_C2 and F1_C3). F1_C1 is a source for transmitting voice calls. F1_C2 is a node for receiving voice calls. F1_C3 is receiver and source of background traffic. This model allows the generation of background traffic and the establishment of links per floor from F1_C1 and F1_C2, which pass through the floor switch, F1SW. VoIP transmitter and receiver nodes (for example F1_C1 and F1_C2) have an infinite capacity and there is no limit to the number of calls generated or received by them. Signaling traffic generated by the *gatekeeper* will be ignored. The analysis and design are based on the worst-case scenario for VoIP traffic. The signaling traffic in which the *gatekeeper* node is involved is generated before the voice call is established and upon its termination. This traffic is relatively limited and small compared to voice traffic. Generally, the *gatekeeper* node does not generate signaling traffic for the duration of a VoIP call for a call that is in progress.

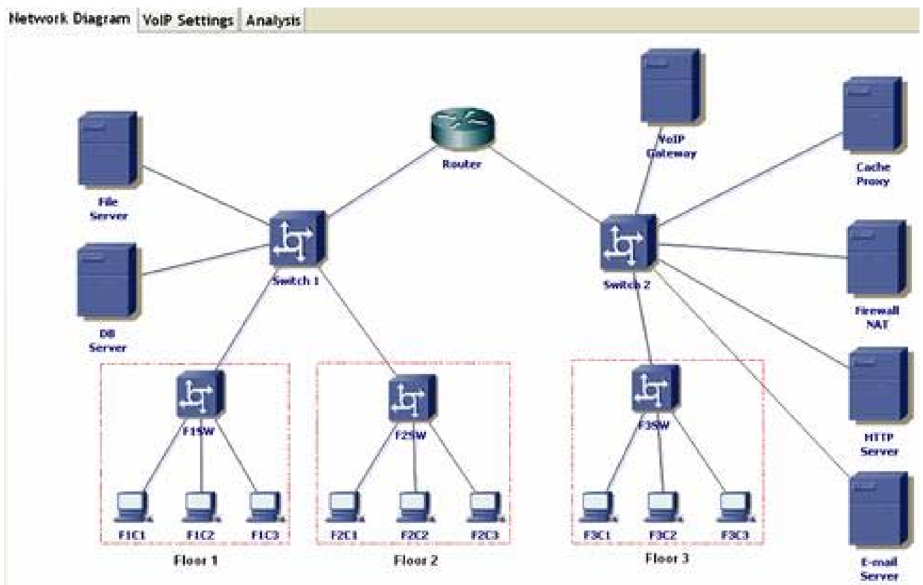
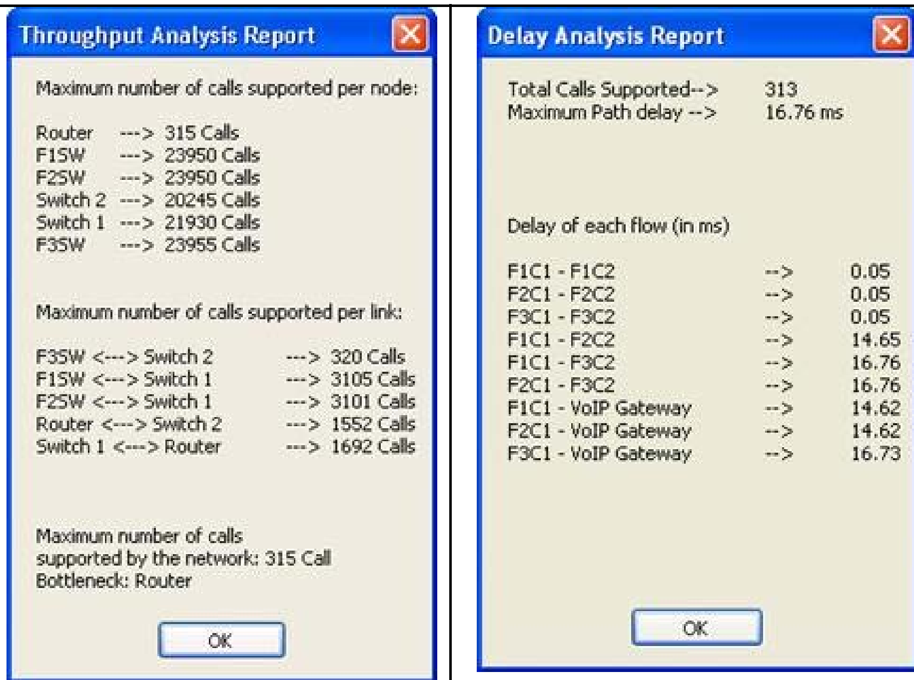


Figure 5. The corresponding network diagram built in the simulator

Figure 6 shows the transfer ratios and delay analyses. Figure 6(a) reports the number of calls that can be supported, based on bandwidth analysis. A total of 315 calls can be accepted for the entire network. In order to identify possible bottlenecks, the report also shows the individual calls that can be supported per node and link. It shows that the router is fully loaded and offering more than 315 calls would require a replacement for it. Figure 6(b) reports the number of calls that can be offered based on the network analysis. 313 calls can be supported, so that the network delay for any VoIP flow does not exceed the defined 80 ms. The figure shows that, for the 313 calls, a network delay of 16.76 ms will be introduced. This means that when another call is added, the network delay limit of 80 ms is exceeded. The report in Figure 6(b) also shows the network delay for each flow. In the given example there were nine VoIP flows. As already specified, the first triplet refers to intra-floor flows. The second triplet relates to inter-floor flows. The third triplet indicates external flows. This information provides insight into the sources of delays, as well as the paths that cause the most delays. As the figure shows, the inter-floor flows F1C1-F3C2 and F2C1-F3C2 have the highest delays because they pass through the router.

Therefore, it can be concluded that the network can support up to 313 end-to-end calls with a delay of 16.76 ms. Network delay (for this network topology) is the dominant factor (more important than throughput) in determining the number of voice calls that can be supported.



(a)	(b)

Figure 6. Transfer Analysis Report (a) and Delay Analysis Report (b)

Steps for homework 1:

1. Start the VoIP Analytical Simulator program.
2. The predefined example opens.
3. Note the network topology.
4. Note the properties of the elements and the defined flows.
5. Note the VoIP settings.
6. Delay analysis transfer analysis reports are generated.
7. Note the elements that generate restrictions in the network.

Steps for homework 2:

1. Imagine a VoIP topology for Transport Faculty.
2. Implement the topology in the VoIP Analytical Simulator program.
3. Configure the properties of the elements.
4. Define VoIP flows.
5. If applicable, modify the VoIP settings as required.
6. Delay analysis transfer analysis reports are generated.
7. Note the elements that generate restrictions in the network.