Voice Models User's Guide

7.0
Contract DASW01 03 D 0008

Disclaimer: As of October 2007, NETWARS was redesignated by the Program Manager Office as the Joint Communications Simulation System (JCSS). JCSS was selected as the new industry name to better reflect the inherent joint communication capabilities of the software. Users should be aware that no software updates were conducted as part of the software name change.

January 7, 2008

Prepared for:
Defense Contracting Command - Washington
Washington, DC 20310-5200

Prepared by:
OPNET Technologies, Inc.
Bethesda, MD 20814-7904

OPNET®
Optimum Network Performance

Contents

List of Figures

List of Tables

1 Overview

Voice devices are defined by their ability to support the transmission of Voice data across many types of networks. Voice refers to the actual data traffic being streamed across a network for communications equipment such as phones or radios. This type of traffic can place significant demands on a particular network, such as:

- High demands on Quality of Service for the network. Users should expect to communicate with each other without interruption or static.

- Large amounts of data on the network as Voice calls typically last for long periods of time. During this period, information must continually be sent.

- Users expect certain types of additional services.

Due to these demands, Voice devices can become complex to use and configure. Also, based on the type of demands by the user, many valid configurations are used and provide different tradeoffs to the end user. An example tradeoff would be using the legacy guaranteed, circuit-based services versus the cost effective, packet-based VoIP services.

Voice devices allow the user to study these types of devices accurately with a high level of fidelity.

1.1 Intended Use

This document is intended for NETWARS users who have a reasonable level of knowledge about voice devices and their use in voice networks. This document is written to help the engineer learn about the configuration of these devices in NETWARS and the types of interfaces, attributes, and statistics available in both the Discrete Event Simulation and Capacity Planning use cases.

This document focuses on the circuit switch, VTC, and gateway aspects of the Voice model suite. The VoIP technologies (i.e., H.323 and SIP) are reserved for other documents. For information on the H.323 models, please refer to the *H.323 Models User's Guide*. For information on the SIP protocol and models, please refer to the *SIP Models User's Guide*.

1.2   Feature Summary

This section provides a list of the main features available in the Voice model suite.

Table 1: Voice Features

Devices	Description	Support
Circuit Switch Terminals	Circuit Switch Terminals provide IER traffic support only. Work with NETWARS tactical circuit switches.	Supported
VTC Terminals	VTC Terminals provide IER traffic support only. Model based on the Circuit Switch technology only.	Supported
VoIP Phones	H.323 and SIP support. Phones support both IERs and Application traffic.	Supported
Tactical (Dynamic) Circuit Switches	Based on tactical circuit switch devices.	Supported
PSTN (Static) Circuit Switches	Supported inside the standard OPNET models. Do not work with the NETWARS circuit switch models or gateways.	Not Supported
Multipoint Control Units (MCUs)	Allows for teleconference support with the Circuit Switch VTC terminals.	Supported
Gateways	Supports the conversion between Circuit Switch, SIP, and H.323 voice technologies. Based on the N.E.T. SHOUTip device.	Supported
SIP Proxy Server	Works with the SIP protocol.	Supported
H.323 Gatekeeper	Based on Cisco IOS Gatekeeper.	Supported
H.323 Border Elements (BEs)	Border Elements will only provide basic RAS signaling and will not include the H.225 Annex G signaling.	Supported
H.323 Multipoint Control Units (MCUs)	Allows for teleconference support in H.323 networks.	Not Supported

1.3   Feature Description

To support the necessary Voice devices, the Voice model suite is an extensive list of models which include circuit switching (CS), voice/video teleconferencing (VTC), and voice over IP (VoIP) technologies. These particular technologies define the various protocols needed to send audio and video over different types of networks. These networks include circuit based, pipe networks, as well as, non-guaranteed, packet-based networks (i.e., IP networks). Typically, in most networks today, a combination of these technologies is used to provide certain quality, cost, or service to the end user.

The Voice model suite provides the ability to accurately study these network conditions by specifying common, real-world end devices (phones), routers/switches, gateways, and other special voice elements. To perform this task, each device contains separate functionality which allows calls to be routed to the destination successfully. Therefore, the configurable parameters of the Voice devices can be divided into four main groups namely, Circuit Switch, VTC, Gateway, and VoIP.

1.4   Limitations/Assumptions

Due to the broad spectrum of protocols available for Voice technologies and based on the NETWARS community needs, some modeling tradeoffs for Voice modeling have been made. These include:

- NETWARS models tactical circuit switching technology in the circuit switch models. This tactical environment is based off the dynamic Flood Search Routing (FSR) protocol. This means that upon call setup, a dynamic handshake occurs between the end phones to setup the complete circuit and reserve the necessary voice channels on the links. No graph creation or static protocol (i.e., PSTN) is modeled in the circuit switch models. For more information, the user can refer to the NETWARS OPNETWORK sessions or contact the NETWARS Help Desk.

- The NETWARS circuit switch models are only able to route Voice or VTC IERs. These models do not support Data IERs or Application traffic.

- The NETWARS Video/Voice Teleconferencing (VTC) models only support a pure circuit switch environment. This means that VTC calls inside NETWARS cannot be routed between the circuit switch and VoIP technologies.

- The NETWARS SIP VoIP protocol is based off of the standard OPNET SIP models. The NETWARS gateways work with these models to create the SIP functionality.

1.5   Model List

The standard NETWARS model library contains a number of node models representing various Voice devices. Those models can be located under the "Node Models" Directory of the **NETWARS_Circuit Switches**, **NETWARS_VTC**, and **NETWARS_VoIP** palettes and include the following models.

The model names are shown with the number of ports available for each port type.

Table 2: Circuit Switch Models

Model Name	Circuit Switch	Phone Switch
Definity Prologic	50	5
DNVT	-	1
DPA	1	2
DPM	1	16
DSS-1, DSS-2, DSS-3	36	5
DTA	1	2
Generic Ckt Switch	32	5
Generic Telephone	-	1
LEN	8	7

NCS	16	5
REDCOM HDX	128	5
REDCOM IGX 1 Shelf	4	5
REDCOM IGX 2 Shelf	8	5
REDCOM IGX 3 Shelf	12	5
REDCOM IGX 4 Shelf	16	5
REDCOM IGX 5 Shelf	20	5
REDCOM IGX 6 Shelf	24	5
REDCOM IGX 7 Shelf	28	5
REDCOM IGX 8 Shelf	32	5
REDCOM IGX 10 Shelf	40	5
REDCOM IGX 16 Shelf	64	5
SB-3865 1 Stack	3	5
SB-3865 2 Stack	6	5
SB-3865 3 Stack	9	5
SEN	13	10
SMU	12	5
TTC-39	30	10
TTC-39A V3	16	5
TTC-39A V4	30	5
TTC-39D	28	5
TTC-39E	6	5
TTC-42	7	5
TTC-46 LEN	9	5
TTC-48 SEN	13	5
TTC-56	19	5

Table 3: VTC Models

Model Name	Circuit Switch	Phone Switch
MCU	-	1
VTC_Terminal	-	1

Table 4: VoIP Models

Model Name	Ethernet	Circuit Switch	Circuit/Serial	ATM
VoIP_Phone	1	-	-	-
SHOUTip	2	8	32	-
Media_Gateway	2	8	-	2
H323_Border_Element	2	-	-	-
H323_Gatekeeper	2	-	-	-
SIP Proxy Server	1	-	-	-

1.6   Reference Documents

- SIP

- RFC 2543. "SIP: Session Initiation Protocol." March 1999.

- RFC 3261. "SIP: Session Initiation Protocol." June 2002.

- H.323

- "H.225.0 – Call signaling protocols and media stream packetization for packet-based multimedia communication systems". Version 5. International Telecommunication Union - Telecommunication Standardization Sector (ITU-T).

- "Understanding H.323 Gatekeepers". Cisco Systems. Document ID 5244. http://www.cisco.com/warp/public/788/voip/understand-gatekeepers.html

1.7   Additional Information

For any additional information not provided in this document, consult the NETWARS Model Development Guide (MDG) or other NETWARS documentation. Information provided in the Model Development Guide includes interfacing with NETWARS models, using NETWARS traffic types, expected packet formats and interrupts, and suggested model development processes. This documentation is provided inside a standard NETWARS installation and can be found in the **<NETWARS Installation Directory>\Documents** directory (where **<NETWARS Installation Directory>** is the directory where NETWARS is installed).

Also, for any questions regarding OPNET specific functionality, consult the standard OPNET documentation. This documentation can be accessed inside NETWARS via the **Help > Documentation > ITGuru Documentation** menu.

Finally, all Voice files can be located under the following directories and sub-directories:

- **<NETWARS Installation Directory>\Sim_Domain\op_models\netwars_std_models\circuit_switch_voice**

- **<NETWARS Installation Directory>\Sim_Domain\op_models\netwars_std_models\voip**

Note that these models and files can be accessed and modified using the OPNET Modeler software.

2 Configuration

One of the advantages of NETWARS is that it provides a great amount of flexibility when configuring network and device models. As the NETWARS Voice suite of models contains a large amount of functionality, additional steps may be needed to configure the necessary functionality for the modeled Voice network.

2.1 Node Configuration

Currently, there are several types of Voice devices available inside the NETWARS palette. To find the available Voice models, open the object palette (**Topology > Open Object Palette**) and find the **NETWARS_Circuit_Switches**, **NETWARS_VTC**, and **NETWARS_VoIP** palettes (shown below).
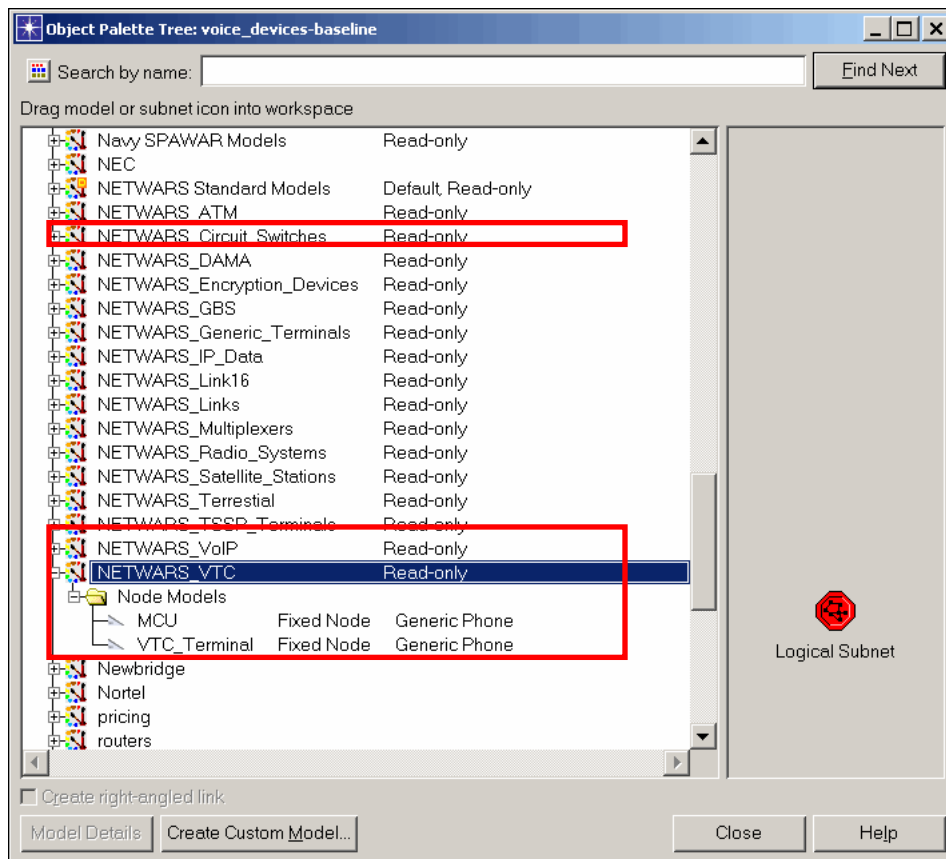
Figure 1: Available Voice Nodes

Once the particular Voice model is found, you can drag and drop the model into the scenario to successfully deploy the model. Once the model is deployed, attributes are available on the device which can be modified to activate certain Voice specific functionality. The following subsections discuss these specific Voice attributes.

2.1.1 Circuit Switch Device Attributes

The Circuit Switch device attributes are all of the attributes which can be accessed by right-clicking on a Circuit Switch device and selecting the **Edit Attributes** menu. The following sections describe the node attributes for each type of device in detail. Any attributes which are not included inside the tables (i.e., "Applications", "CPU", etc.) refer to OPNET standard attributes. Please refer to the OPNET documentation for further details on these attributes.

2.1.1.1   Circuit Switch Terminal Parameters

These parameters refer to any node attributes which are assigned to the Circuit Switch terminals. Circuit Switch terminals are devices which act as end stations (i.e., phones) inside the Circuit Switch network. Example devices which fall under this specification are the **DNVT** and **Generic Telephone** models. The following table illustrates the attributes with descriptions needed to properly configure the Circuit Switch terminals.

Figure 2: Circuit Switch Terminal Parameters

Table 5: Circuit Switch Terminal Parameters

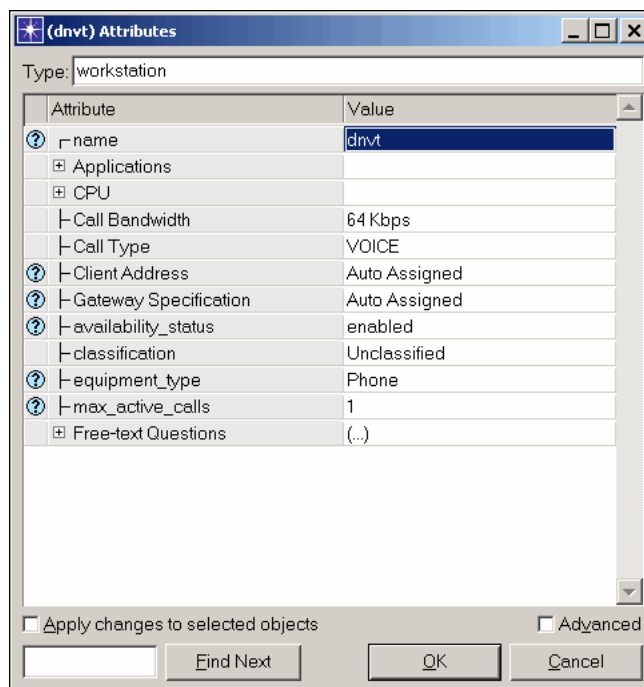| Attributes	Description
Call Bandwidth	This attribute specifies the bandwidth (in bps) which will be required for each voice call that uses this terminal (as a source or destination terminal). When a voice call is being setup, all voice channels along the route of the call must support the bandwidth which is specified by this attribute. If so, then any available voice channels on the links will be reserved for the duration of the call. If not, then the voice call will be failed due to a lack of bandwidth. In cases where the source and destination terminals have different call bandwidth values, the lower value of the two terminals will be used. The default value is 64 Kbps.
Call Type	This attribute specifies the type of call which will be accepted by this terminal during simulation. If the value is set to "VOICE," then only Voice IERs will be able to use this terminal. If the value is set to "VTC," then only VTC IERs will be able to use this terminal. The default for this attribute is the "VOICE" value and should not be changed.
Gateway Specification	This attribute specifies the gateway this phone will signal if the destination circuit switch terminal cannot be found. If a gateway exists in the network, the hierarchical name of that gateway will be available as an option for this attribute. If the user selects the "Auto Assigned" option, the circuit switch terminal will dynamically lookup all gateways available in the simulation and will randomly pick a gateway. The default for this attribute is the "Auto Assigned" value.
Max Active Calls	This attribute specifies the maximum number of simultaneous calls on this circuit switch terminal. This allows the terminal to act as several terminals instead of having the user deploy many terminals into the scenario. The default for this attribute is '1' meaning that this terminal will act as a single terminal.
Availability Status	A NETWARS standard attribute, denotes whether the node is available to receive IER traffic during a simulation.
Classification	A NETWARS standard attribute, sets the classification level (i.e., Unclassified, Classified, etc.) of this particular circuit switch terminal.
Equipment Type	A NETWARS standard attribute, sets the device type for NETWARS specific functionality. This attribute should always be set to "Phone."

2.1.1.2   Circuit Switch Parameters

These parameters refer to any node attributes which are assigned to the Circuit Switch devices. Circuit Switch devices are devices which act as switches (i.e., switch/routers) inside the Circuit Switch network. Example devices which fall under this specification are the **LEN**, **SEN**, and **TTC-39** models. The following table illustrates the attributes with descriptions needed to properly configure the Circuit Switch devices.
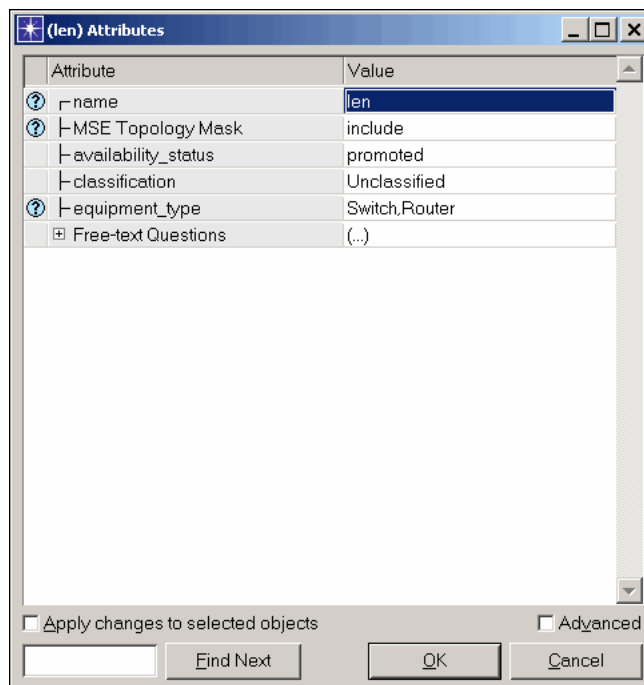
Figure 3: Circuit Switch Parameters

Table 6: Circuit Switch Parameters

Attributes	Description
MSE Topology Mask	This attribute specifies that this device should be included inside the Circuit Switch network. This attribute should always be set to "include."
Availability Status	A NETWARS standard attribute, denotes whether the node is available to receive IER traffic during a simulation.
Classification	A NETWARS standard attribute, sets the classification level (i.e., Unclassified, Classified, etc.) of this particular circuit switch device.
Equipment Type	A NETWARS standard attribute, sets the device type for NETWARS specific functionality. This attribute should always be set to "Switch, Router."

2.1.2 VTC Device Attributes

The VTC device attributes are all of the attributes which can be accessed by right-clicking on a VTC device and selecting the **Edit Attributes** menu. VTC devices are devices which act as VTC end stations (i.e., terminals) inside the Circuit Switch network. Example devices which fall under this specification are the **VTC_Terminal** and **MCU** models. The following table illustrates the attributes with descriptions needed to properly configure the VTC devices.

Note that as the VTC devices are based on the NETWARS Circuit Switch models, these type of models are deployed with other circuit switch devices (such as circuit switches). To configure the circuit switch models, refer to the section above.

Any attributes which are not included inside the tables (i.e., "Applications", "CPU", etc.) refer to OPNET standard attributes. Please refer to the OPNET documentation for further details on these attributes.
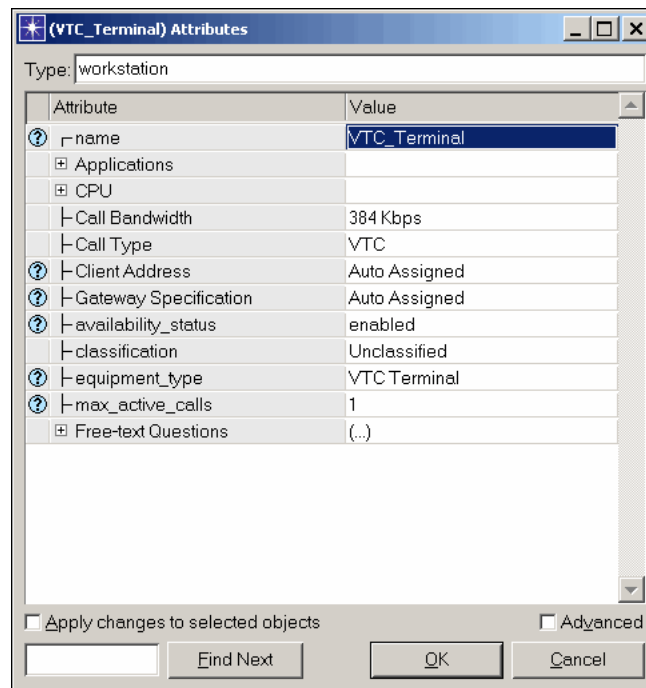

Figure 4: VTC Device Parameters

Table 7: VTC Device Parameters

Attributes	Description
Call Bandwidth	This attribute specifies the bandwidth (in bps) which will be required for each VTC call that uses this terminal (as a source or destination terminal). When a VTC call is being setup, all voice channels along the route of the call must support the bandwidth which is specified by this attribute. If so, then any available voice channels on the links will be reserved for the duration of the call. If not, then the voice call will be failed due to a lack of bandwidth. In cases where the source and destination terminals have different call bandwidth values, the lower value of the two VTC terminals will be used. The default value is 384 Kbps.
Call Type	This attribute specifies the type of call which will be accepted by this terminal during simulation. If the value is set to "VOICE," then only Voice IERs will be able to use this terminal. If the value is set to "VTC," then only VTC IERs will be able to use this terminal. The default for this attribute is the "VTC" value and should not be changed.
Gateway Specification	This attribute specifies the gateway this VTC terminal will signal if the destination VTC terminal cannot be found. If a gateway exists in the network, the hierarchical name of that gateway will be available as an option for this attribute. If the user selects the "Auto Assigned" option, the VTC terminal will dynamically lookup all gateways available in the simulation and will randomly pick a gateway. The default for this attribute is the "Auto Assigned" value.
Max Active Calls	This attribute specifies the maximum number of simultaneous calls on this VTC terminal. This allows the terminal to act as several terminals instead of having the user deploy many terminals into the scenario. The default for this attribute is '1' meaning that this terminal will act as a single terminal.
Availability Status	A NETWARS standard attribute, denotes whether the node is available to receive IER traffic during a simulation.
Classification	A NETWARS standard attribute, sets the classification level (i.e., Unclassified, Classified, etc.) of this particular VTC terminal.
Equipment Type	A NETWARS standard attribute, sets the device type for NETWARS specific functionality. This attribute should always be set to "VTC Terminal."

2.1.3 VoIP Device Attributes

The VoIP device attributes are all of the attributes which can be accessed by right-clicking on a VoIP device and selecting the **Edit Attributes** menu. The following sections describe the node attributes for each type of device in detail. Any attributes which are not included inside the tables (i.e., "Applications", "CPU", etc.) refer to OPNET standard attributes. Please refer to the OPNET documentation for further details on these attributes.

2.1.3.1 VoIP Phone Attributes

The following table describes the VoIP Phone node attributes in detail. Any relevant compound node attributes are left for subsequent tables as compound attributes contain another layer of attributes. All other attributes refer to the routing capabilities of this H.323 terminal and are out of scope for this document.
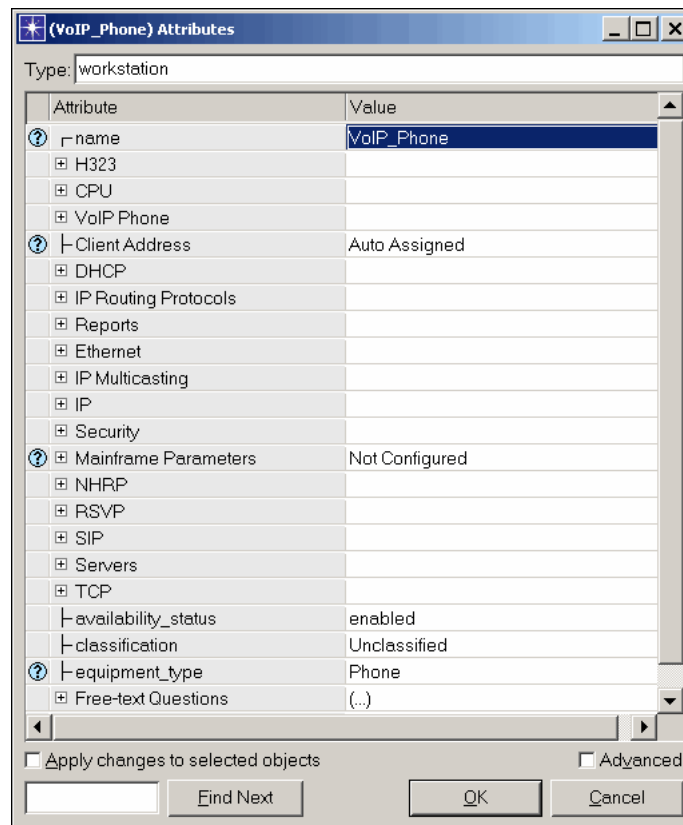
Figure 5: VoIP Phone Node Attributes

Table 8: VoIP Phone Node Attributes

Attributes	Description
Availability Status	A NETWARS standard attribute, denotes whether the node is available to receive IER traffic during a simulation.
Classification	A NETWARS standard attribute, sets the classification level (i.e., Unclassified, Classified, etc.) of this particular VoIP phone.
Equipment Type	A NETWARS standard attribute, sets the device type for NETWARS specific functionality. This attribute should always be set to "Phone."

VoIP Phone Parameters

As the **VoIP_Phone** node model allows configuration of both H.323 and SIP protocol voice calls, the **VoIP Phone** compound attribute was added so the user could configure attributes which would apply to both protocols. The following table illustrates the attributes with descriptions needed to properly configure the **VoIP_Phone** device.
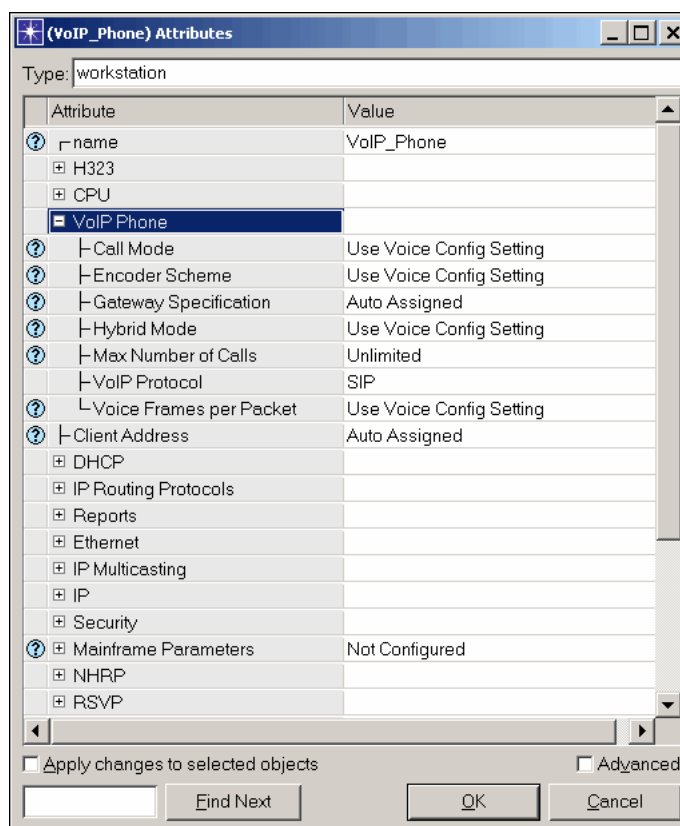

Figure 6: VoIP Phone Compound Attribute Parameters

Table 9: VoIP Phone Compound Attribute Parameters

Attributes	Description
Call Mode	This attribute specifies the different call modes which are available on the VoIP Phone. If the user selects "Signaling and Data," then both the VoIP protocol call signaling and the voice data packets will traverse the network. If the user selects "Signaling Only," then only the VoIP protocol call signaling will occur. In this case, no voice data packets will be sent. If the user selects "Use Voice Config Setting," then this VoIP Phone will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.
Encoder Scheme	This attribute specifies the encoder scheme which will be used for all calls which use this particular VoIP Phone. The encoder scheme specifies the information about the voice data packets sent during the call. Such information includes packet intervals and packet size. The available encoder schemes are based on standard VoIP phone technologies. If the user selects "Use Voice Config Setting," then this VoIP Phone will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.
Gateway Specification	This attribute specifies the gateway this phone will signal if the destination VoIP phone cannot be found. If a gateway exists in the network, the hierarchical name of that gateway will be available as an option for this attribute. If the user selects the "Auto Assigned" option, the VoIP phone will dynamically lookup all gateways available in the simulation and will randomly pick a gateway.
Hybrid Mode	This attribute allows the user to choose between the use of hybrid or explicit voice data packets during a simulation. If the user chooses the "Hybrid Packets" option, the VoIP phone will load the network using hybrid packets for all voice data packets. This option will increase the speed of the simulation with the tradeoff of losing some statistics (such as jitter and end-to-end delay). If the user picks the "Explicit Packets" option, the VoIP Phone will load the network using an explicit packet for each voice data packet. This slows the simulation down but provides a detailed analysis. If the user selects "Use Voice Config Setting," then this VoIP Phone will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.
Max Number of Calls	This attribute specifies the maximum number of simultaneous calls on this VoIP Phone. This allows the phone to act as several phones instead of having the user deploy many phones onto the scenario. If the user selects the "Unlimited" option, there is no limit to the number of calls placed on the VoIP phone at a single time.
VoIP Protocol	This attribute specifies the call signaling protocol which will be used to setup/teardown a voice call on this phone. If this attribute is set to "SIP," the calling signaling protocol will be the Session Initiation Protocol (SIP). If the attribute is set to "H.323," the protocol used will be the H.323 umbrella protocol.
Voice Frames per Packet	This attribute specifies the number of voice frames per packet. This attribute coincides with the encoder scheme attribute above and further specifies information about the voice data packet. This attribute affects the amount of data which is included inside a voice data packet. If the user selects "Use Voice Config Setting," then this VoIP Phone will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.

SIP and H.323 Parameters

As the **VoIP_Phone** node model allows configuration of both H.323 and SIP protocol voice calls, additional compound attributes are needed to configure these protocols (shown below). However, as these attributes are based on protocol standards, these attributes are out of scope for this document.

For information on how to configure the SIP protocol, please refer to the *SIP Models User's Guid*e.

For information on how to configure the H.323 protocol, please refer to the *H.323 Models User's Guide*.

Figure 7: VoIP Phone SIP and H.323 Parameters

2.1.3.2   Other VoIP Device Attributes

As all the other models inside the NETWARS VoIP suite are specific to the H.323 and SIP protocols, these models (i.e., the **H323_Border_Element**, **H323_Gatekeeper**, and **SIP Proxy Server** models) are out of scope for this document.

For information on how to configure the SIP protocol, please refer to the *SIP Models User's Guid*e.

For information on how to configure the H.323 protocol, please refer to the *H.323 Models User's Guide*.

2.1.4 Gateway Device Attributes

The Gateway device attributes are all of the attributes which can be accessed by right-clicking on the **Media_Gateway** or **SHOUTip** devices and selecting the **Edit Attributes** menu. The following table describes the node attributes in detail. Any relevant compound node attributes are left for subsequent tables as compound attributes contain another layer of attributes. All other attributes refer to the routing capabilities of this gateway and are out of scope for this document. Please refer to the standard OPNET documentation for more information.
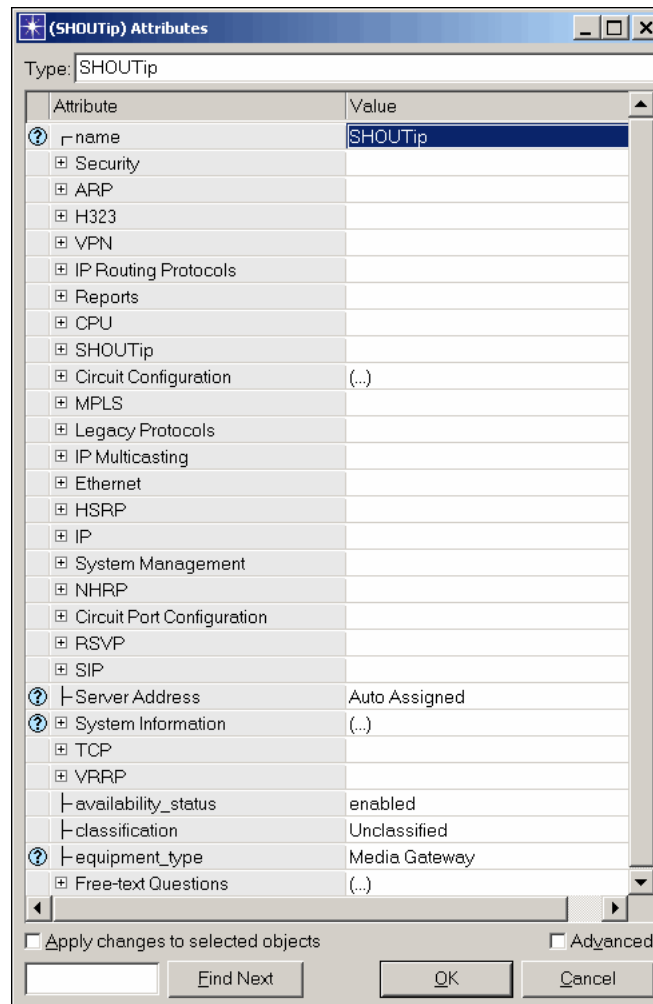
Figure 8: Gateway Attributes

Table 10: Gateway Attributes

Attribute	Description
Availability Status	A NETWARS standard attribute, denotes whether the node is available to receive IER traffic during a simulation.
Classification	A NETWARS standard attribute, sets the classification level (i.e., Unclassified, Classified, etc.) of this particular gateway.
Equipment Type	A NETWARS standard attribute, sets the device type for NETWARS specific functionality. This attribute should always be set to "Media Gateway."

2.1.4.1 Media Gateway or SHOUTip Parameters

As the **Media_Gateway** or **SHOUTip** node models allow configuration of both H.323 and SIP protocol voice calls, the **Media Gateway** or **SHOUTip** compound attributes were added so the user could configure attributes which would apply to both protocols. The following table illustrates the attributes with descriptions needed to properly configure the gateway.
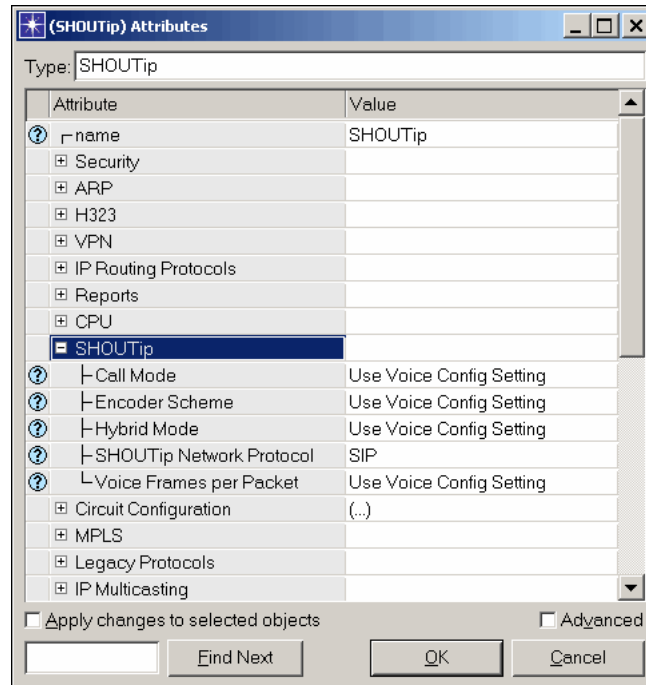
Figure 9: Media Gateway or SHOUTip Parameters

Table 11: Media Gateway or SHOUTip Parameters

Attributes	Description
Call Mode	This attribute specifies the different call modes which are available on the gateway. If the user selects "Signaling and Data," then both the VoIP call signaling and the voice data packets will traverse the network. If the user selects "Signaling Only," then only the VoIP call signaling will occur. In this case, no voice data packets will be sent. If the user selects "Use Voice Config Setting," then this gateway will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.
Encoder Scheme	This attribute specifies the encoder scheme which will be used for all calls which use this particular gateway. The encoder scheme specifies the information about the voice data packets sent during the call. Such information includes packet intervals and packet size. The available encoder schemes are based on standard gateway technologies. If the user selects "Use Voice Config Setting," then this gateway will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.
Hybrid Mode	This attribute allows the user to choose between the use of hybrid or explicit voice data packets during a simulation. If the user chooses the "Hybrid Packets" option, the gateway will load the network using hybrid packets for all voice data packets. This

	option will increase the speed of the simulation with the tradeoff of losing some statistics (such as jitter and end-to-end delay). If the user picks the "Explicit Packets" option, the gateway will load the network using an explicit packet for each voice data packet. This slows the simulation down but provides a detailed analysis. If the user selects "Use Voice Config Setting," then this gateway will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.
SHOUTip Network Protocol	This attribute specifies the call signaling protocol which will be used to setup/teardown a voice call on this gateway. If this attribute is set to "SIP," the calling signaling protocol will be the Session Initiation Protocol (SIP). If the attribute is set to "H.323," the protocol used will be the H.323 umbrella protocol.
Voice Frames per Packet	This attribute specifies the number of voice frames per packet. This attribute coincides with the encoder scheme attribute above and further specifies information about the voice data packet. This attribute affects the amount of data which is included inside a voice data packet. If the user selects "Use Voice Config Setting," then this gateway will use the setting for the matching attribute located in the Voice Config node. Note that the Voice Config node is located in the Configuration OPFAC.

2.1.4.2   SIP and H.323 Parameters

As the **Media_Gateway** or **SHOUTip** node models allow configuration of both H.323 and SIP protocol voice calls, additional compound attributes are needed to configure these protocols (shown below). However, as these attributes are based on protocol standards, these attributes are out of scope for this document.

For information on how to configure the SIP protocol, please refer to the *SIP Models User's Guid*e.

For information on how to configure the H.323 protocol, please refer to the *H.323 Models User's Guide*.
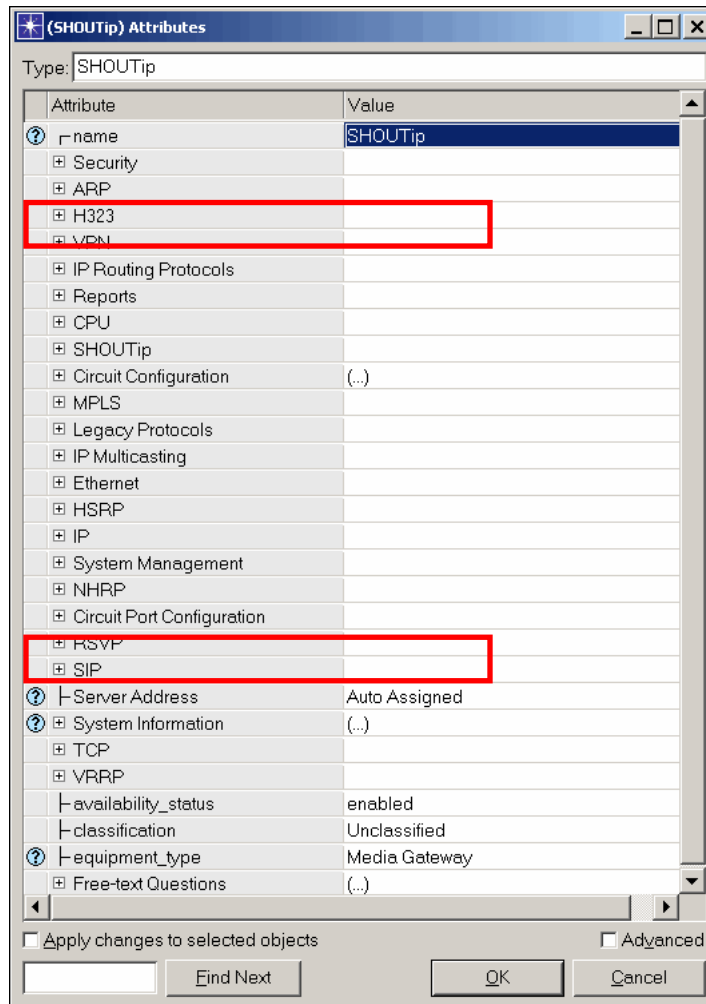
Figure 10: Gateway SIP and H.323 Parameters

2.2 Link Configuration

As the Voice devices are used in conjunction with the several types of networks, these devices contain many different types of ports. Therefore, the link possibilities for these devices are not limited to a single technology. Currently, there are two ways to deploy links inside NETWARS:

1. Link Deployment Wizard (**Topology > Link Operations > Deploy Link** or **Ctrl +
   L**)

2. Object Palette (**Topology > Open Object Palette**)

Currently, the easiest way to connect a Voice device is to use the Link Deployment Wizard (LDW). To accomplish a successful configuration, use the following steps (see the matching LDW figure below):

1. Select the two devices which need to be connected and press **Ctrl + L**.

2. When the Link Deployment Wizard appears, select an available link technology from the menu (if an error message appears, the Voice device is unable to connect to this type of device).

3. Select the link which is appropriate for this technology. For example, if the specified technology is Ethernet then the 10BaseT, 100BaseT, and 1000BaseT links are available. The following list shows the applicable technologies for each type of device:

   a. Circuit Switch Terminals: Phone Switch links

   b. Circuit Switch Devices: Circuit Switch and Phone Switch links

   c. VTC Devices: Phone Switch links

   d. VoIP Phone and Devices: Ethernet (10BaseT, 100BaseT, and 1000BaseT) links

   e. Gateway: Circuit Switch, Ethernet (10BaseT, 100BaseT, and 1000BaseT), ATM (SONET OC1, SONET OC3, SONET OC12, etc.), and Circuit/Serial links

4. Check to make sure the ports on both the connecting devices are acceptable. NETWARS contains a certain naming convention which will be placed on ports which correspond to certain link types. The following list shows the applicable links with the matching port naming convention.

   a. Phone Switch (ckt_pt_#)

   b. Circuit Switch (dtg_pt_# or cs_pt_#)

   c. Ethernet (eth_tx_#_#)

   d. ATM (atm_tx_#_#)

   e. Circuit/Serial (cir_pt_#)

5. Change the bandwidth on the link. Selecting an initial link (i.e., 10BaseT, 100BaseT, etc.) will set a default bandwidth in the wizard but you can select other bandwidth values as necessary.
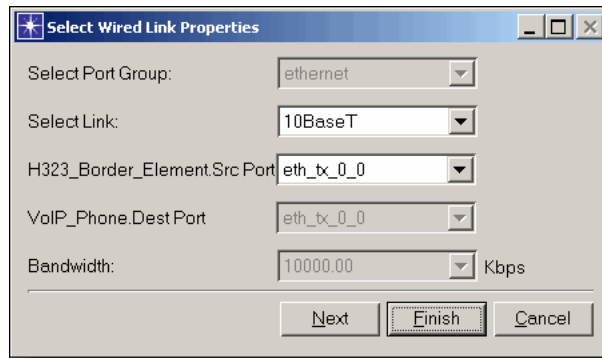
Figure 11: Link Deployment Wizard

Refer to the *NETWARS User's Manual* for further information on deploying links using the above methods.

2.3   Voice Channel Configuration

The circuit switch models in NETWARS provide the ability to configure voice channels on circuit switch links. Voice channels are used by the circuit switch models to reserve the necessary bandwidth on link for a particular voice call. For each circuit switch link on the path from the source phone to the destination phone, there must be a voice channel available with at least the same amount of bandwidth as the voice call. Note that voice channels can be reserved for both Voice and VTC calls (IERs).

To configure voice channels inside NETWARS, the user must modify the characteristics of the links. Depending on whether the link is a Circuit Switch or Phone Switch link, the process is different. Shown below are the steps necessary to configure voice channels on each type of link.

2.3.1 Configuring Voice Channels on Circuit Switch Links

Typically, Circuit Switch links are the links defined between circuit switch devices (i.e., **LEN**, **SEN**, **TTC-39**, etc.). Also, the Circuit Switch link is used when connecting a circuit switch device to a gateway. To modify the voice channels on these links, use the following steps:

1. Right-click on the Circuit Switch link and select **Edit NETWARS Attributes**.
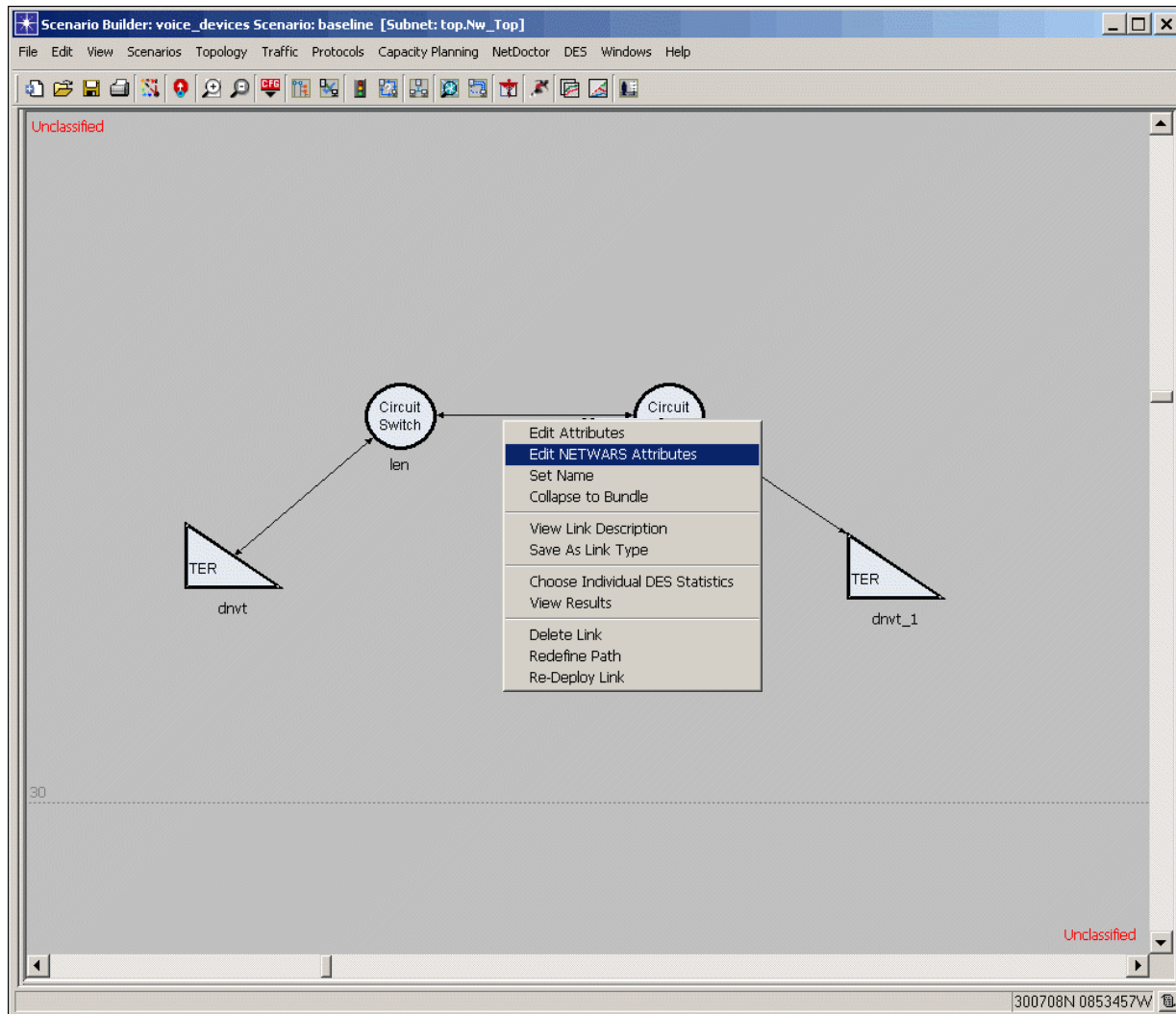
Figure 12: Editing Circuit Switch Link Attributes

2. When the **Wired Link Attributes** dialog box appears, modify the **Number of Voice Channels** and **Channel Size for Voice** attributes. The Number of Voice Channels attribute will specify the number of simultaneous calls which can be present on that link. The Channel Size for Voice attribute will specify the size (in Kbps) of each voice channel on that particular link. For a voice call to use an available channel during simulation, the Channel Size for Voice attribute must be at least equal in size to the **Call Bandwidth** attribute on the Circuit Switch or VTC phone. In cases where VTC is used, the **Call Bandwidth** can be spread across six different channels. Note that the **Remaining Data Bandwidth** attribute shows the remaining bandwidth available on the link. This is calculated by taking the total bandwidth of the link and subtracting the bandwidth used for each voice channel (i.e., **Remaining Bandwidth = Total Bandwidth – (Number of Voice Channels * Voice Channel Size)**). The remaining bandwidth value should never be a negative value.

3. Click OK on the dialog box to accept the changes.
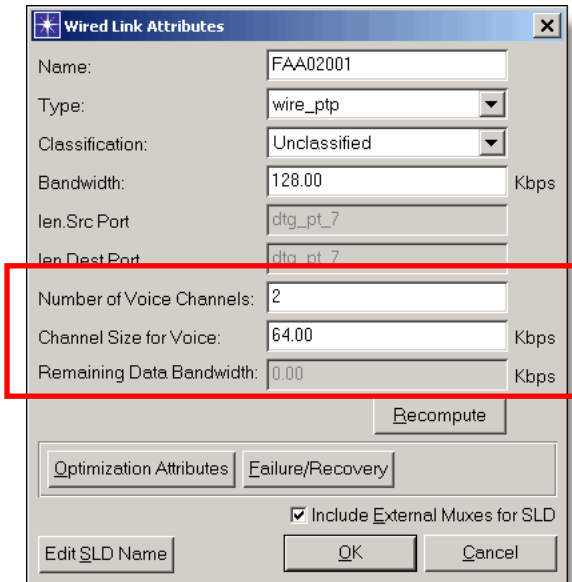
Figure 13: Setting Voice Channels on Circuit Switch Links

2.3.2 Configuring Voice Channels on Phone Switch Links

Phone Switch links are the links defined between a circuit switch device (i.e., **LEN**, **SEN**, **TTC-39**, etc.) and a circuit switch or VTC terminal (i.e., **DNVT**, **VTC_Terminal**, etc.). As this link is always connecting a single end terminal to the border circuit switch device, this link should always have the same number of voice channels as the amount specified by the phone. In typical cases, an end terminal would only have a single voice channel going out on the link connecting it to the circuit switch network. However, in NETWARS, an end terminal can represent many end terminals (so the user does not have to deploy several terminal devices into a scenario). Therefore, to avoid mis-configuration, the Phone Switch link will use the same number of voice channels as the number of users on the phone. To modify the voice channels on these links, use the following steps:

1. Right-click on the end terminal (i.e., a **DNVT**, **VTC_Terminal**, etc.) connecting to the Phone Switch link and select **Edit Attributes**.
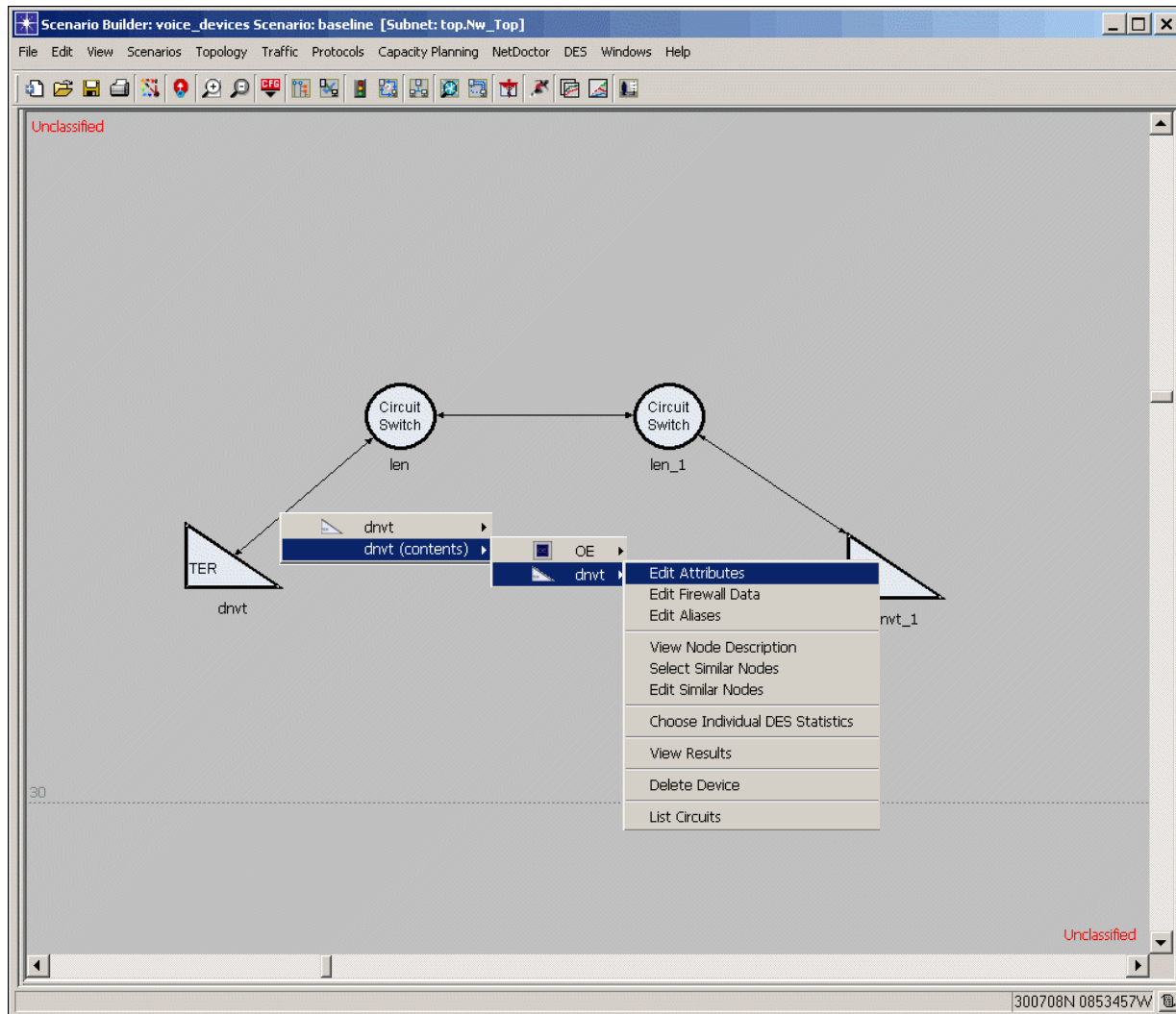
Figure 14: Selecting Phone Switch Link Attributes

2. When the **Edit Attributes** dialog box appears, modify the **Call Bandwidth** and **Max Active Calls** attributes. The Call Bandwidth attribute will specify the size (in Kbps) of each voice call which uses that particular end terminal. Note that during a simulation, the lower Call Bandwidth between the source and destination terminals will be used when reserving voice channels. The Max Active Calls attribute will specify the number of simultaneous calls which can be present on that end terminal.

3. Click OK on the dialog box to accept the changes.
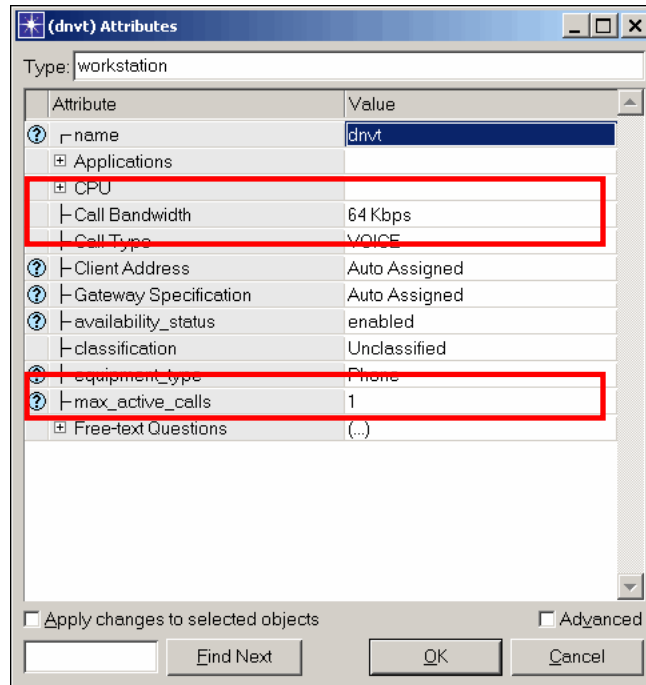
Figure 15: Editing Phone Switch Link Attributes

4. Once the end terminal attributes have been changed for the link, right-click on the Phone Switch link and select **Edit NETWARS Attributes**. When the **Wired Link Attributes** dialog box appears, the user can validate the changes made in Steps 2 and 3. The **Number of Voice Channels** attribute will correspond to the **Max Active Calls** attribute. The **Channel Size for Voice** attribute will correspond to the **Call Bandwidth** attribute on the Circuit Switch or VTC phone. In cases where VTC is used, the **Call Bandwidth** can be spread across six different channels. Note that the **Remaining Data Bandwidth** attribute shows the remaining bandwidth available on the link. This is calculated by taking the total bandwidth of the link and subtracting the bandwidth used for each voice channel (i.e., **Remaining Bandwidth = Total Bandwidth – (Number of Voice Channels * Voice Channel Size)**). The remaining bandwidth value should never be a negative value.

Note that the attributes inside the Wired Link Attributes dialog box cannot be modified. The user must always change the end terminal configuration to change the voice channels on this link. Again, this is by design and avoids mis-configuration inside the scenario.
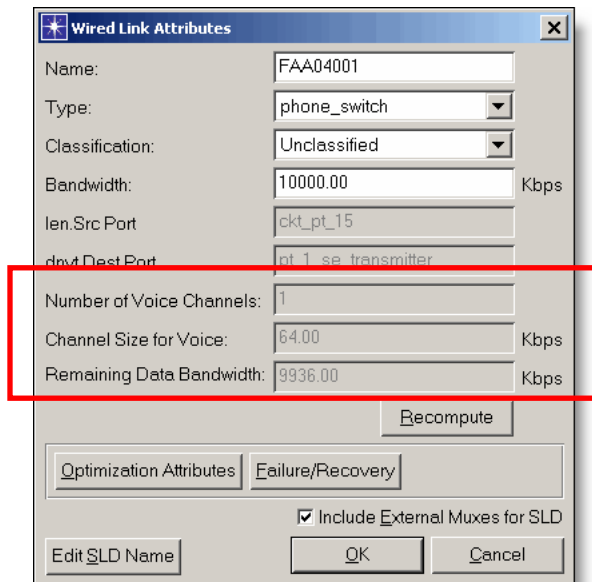
Figure 16: Setting Voice Channels on Phone Switch Links

2.3.3 Matching Encoder Schemes to Voice Channels

When configuring a scenario which contains calls that transfer between the Circuit Switch and VoIP domains, the user should be aware of the **Encoder Scheme** attribute. This attribute is placed on the VoIP phone and specifies the information about the voice data packets sent during the call on the IP network. However, when converting between a VoIP and Circuit Switch network, the gateway must be able to translate the voice packets on the IP network into the same amount of data on the voice channel (or vice versa). Therefore, a translation is made between the encoder scheme used on the VoIP phone and the voice channel connected into the gateway during call setup. The following list shows the available encoder schemes for the VoIP technology and the matching bandwidth which will be reserved inside the Circuit Switch network (in Kbps).

- G.711 – 64 Kbps

- G.711 (Silence Suppression Enabled) – 64 Kbps

- G.723.1 – 24 Kbps

- G.723.1 (Silence Suppression Enabled) – 24 Kbps

- G.726 ADPCM – 32 Kbps

- G.726 ADPCM (Silence Suppression Enabled) – 32 Kbps

- G.727 – 32 Kbps

- G.727 (Silence Suppression Enabled) – 32 Kbps

- G.729 CS-ACELP – 8 Kbps

- G.729 CS-ACELP (Silence Suppression Enabled) – 8 Kbps

Note that if the voice channels inside the Circuit Switch network are not large enough to handle the VoIP call, then the call will fail during setup (as the voice channels cannot be fully reserved). Conversely, if the IP network cannot handle the amount of packets created from the Circuit Switch call, the call will setup but will eventually fail when the quality of service reaches a certain limit.
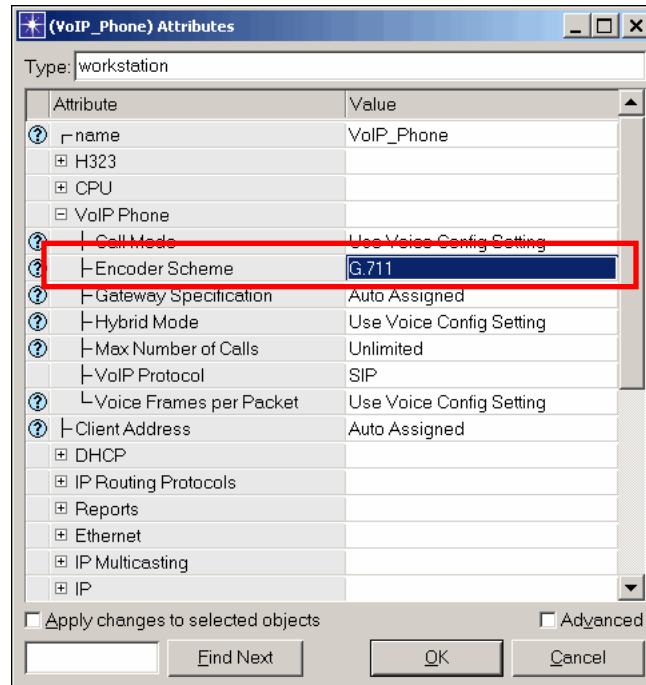
Figure 17: Setting the Encoder Scheme Attribute

3   Usage

The Voice suite of models allows the user to model networks which work with both Discrete Event Simulations and Capacity Planner in NETWARS. Based on the network required, there are several types of Voice devices which provide the necessary functionality. Various types of statistics measure the performance in the Node, Port, or Global level. The figure below shows a simple example network with Voice devices. This example network shows the user a simple setup which can be used to help understand the basics of Voice modeling. Also, the example is intended to demonstrate some of the functionality which is explained in the later sections of this document.

The scenario shown below is a common scenario which includes a Circuit Switch network connected, through a Promina backbone, to a VoIP (H.323) Network. To recreate the scenario, use the following steps:

1. Create a new NETWARS project (**File > New Project…**).

2. Create two OPFACs in the scenario by right-clicking in the workspace and selecting **Create New OPFAC**. Rename the OPFACs to "Circuit_Switch_Network" and "VoIP_Network" by right-clicking on a created OPFAC and selecting **Edit Attributes**.

3. Drill into the "Circuit_Switch_Network" OPFAC (by double clicking on the OPFAC) and deploy 3 LEN, 2 DNVT, and 1 Promina devices using the Object Palette (**Topology > Open Object Palette**). The layout should look similar to the "View of the Circuit_Switch_Network OPFAC" figure below (minus the links).

4. Deploy links between the different devices in the scenario using the Link Deployment Wizard. To do this, select two devices and press **Ctrl + L**. Use the default settings when the wizard appears. The layout should look similar to the "View of the Circuit_Switch_Network OPFAC" figure below.

5. Right-click into the workspace and select **Return to Parent Organization**.

6. Drill into the "VoIP_Network" OPFAC (by double clicking on the OPFAC) and deploy 3 Cisco_4500, 2 VoIP_Phone, 1 SHOUTip, 1 len, and 1 Promina devices using the Object Palette (**Topology > Open Object Palette**). The layout should look similar to the "View of the VoIP_Network OPFAC" figure below minus the links.

7. Deploy links between the different devices in the scenario using the Link Deployment Wizard. To do this, select two devices and press **Ctrl + L**. Use the default settings when the wizard appears. The layout should look similar to the "View of the VoIP_Network OPFAC" figure below.

8. For each VoIP Phone inside the "VoIP_Network" OPFAC, change the **VoIP Phone > VoIP Protocol** attribute to the **H.323** value. To do this, right-click on the device and select **Edit Attributes**.

9. For the SHOUTip device inside the "VoIP_Network" OPFAC, change the **SHOUTip > SHOUTip Network Protocol** attribute to the **H.323** value. To do this, right-click on the device and select **Edit Attributes**.

10. Right-click into the workspace and select **Return to Parent Organization**.

11. Select the two OPFACs in the scenario and use the Link Deployment Wizard to deploy a link between the two Promina devices.

12. Select the two OPFACs in the scenario and use the Generic Circuit Wizard (**Ctrl + D**) to connect a circuit between the Promina devices. Use the defaults for the circuit when the wizard appears.

13. Specify two IERs inside the scenario by using the **Traffic > IERs > Specify IERs** menu. The two IERs should have the following attributes:

    - IER #1: Producer (Circuit_Switch_Network), Consumer (VoIP_Network), Traffic Type (Voice), Size (20), Average Interarrival (100), Distribution Type (CONSTANT).

    - IER #2: Producer (VoIP_Network), Consumer (Circuit_Switch_Network), Traffic Type (Voice), Size (50), Average Interarrival (60), Distribution Type (CONSTANT).
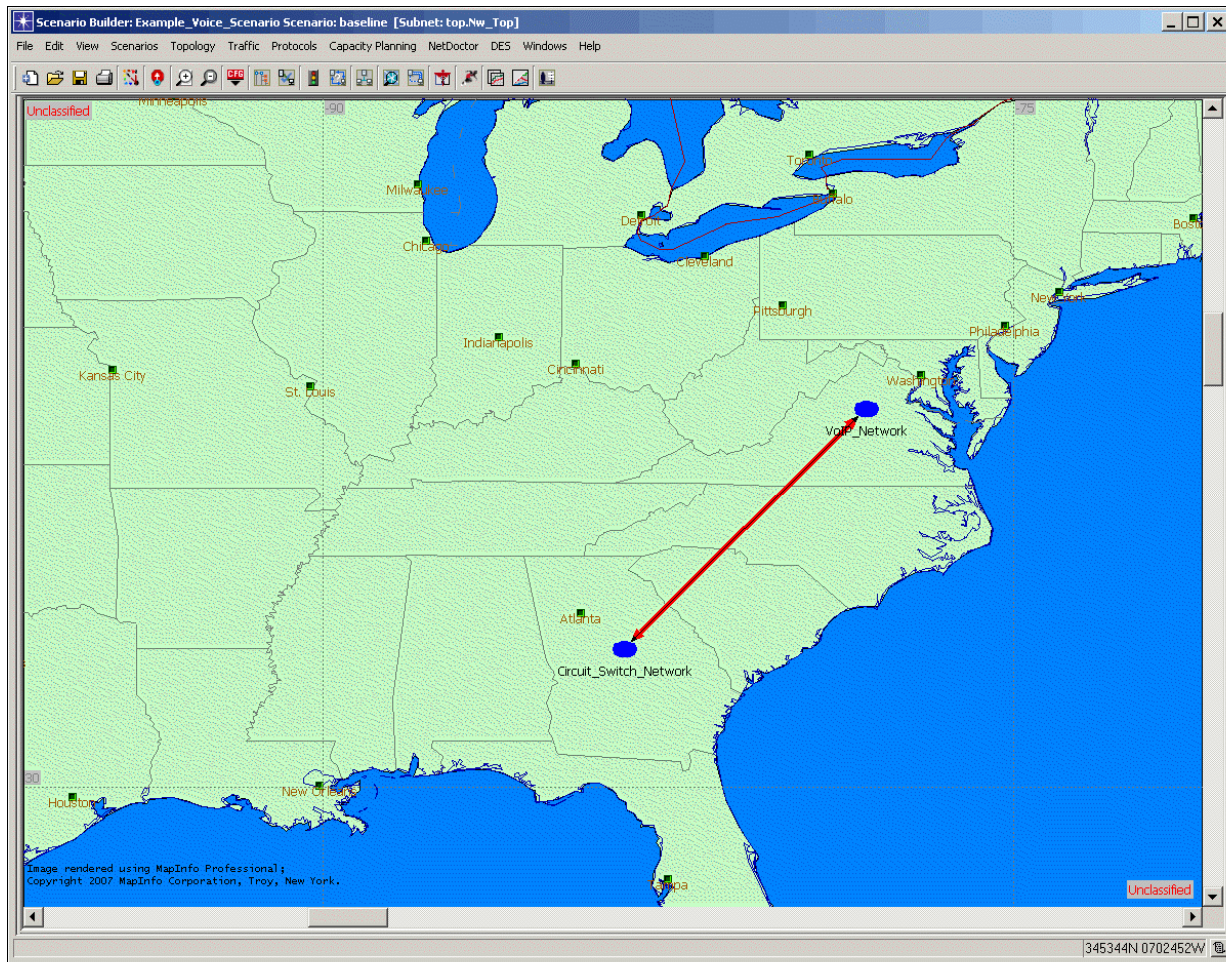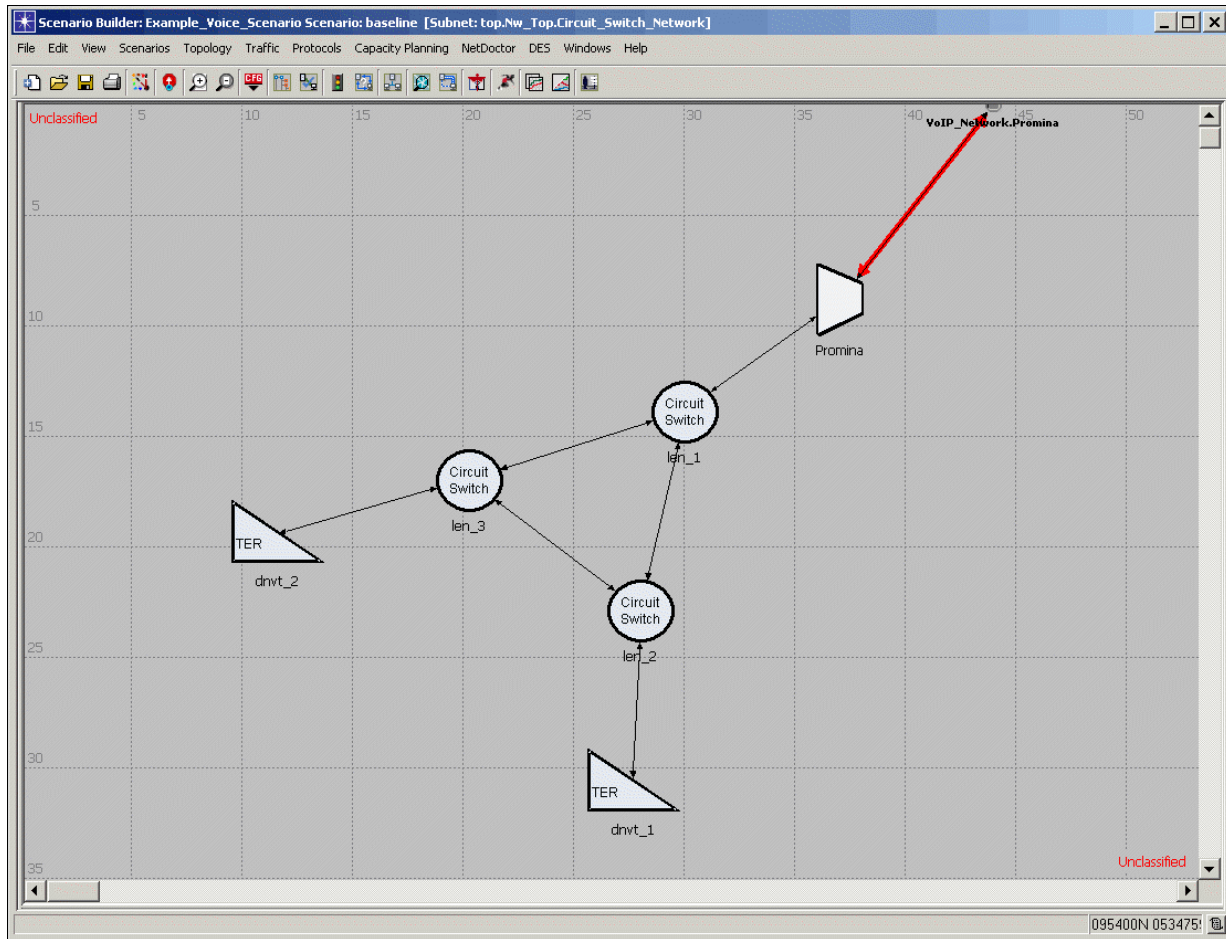
Figure 18: Example Voice Network

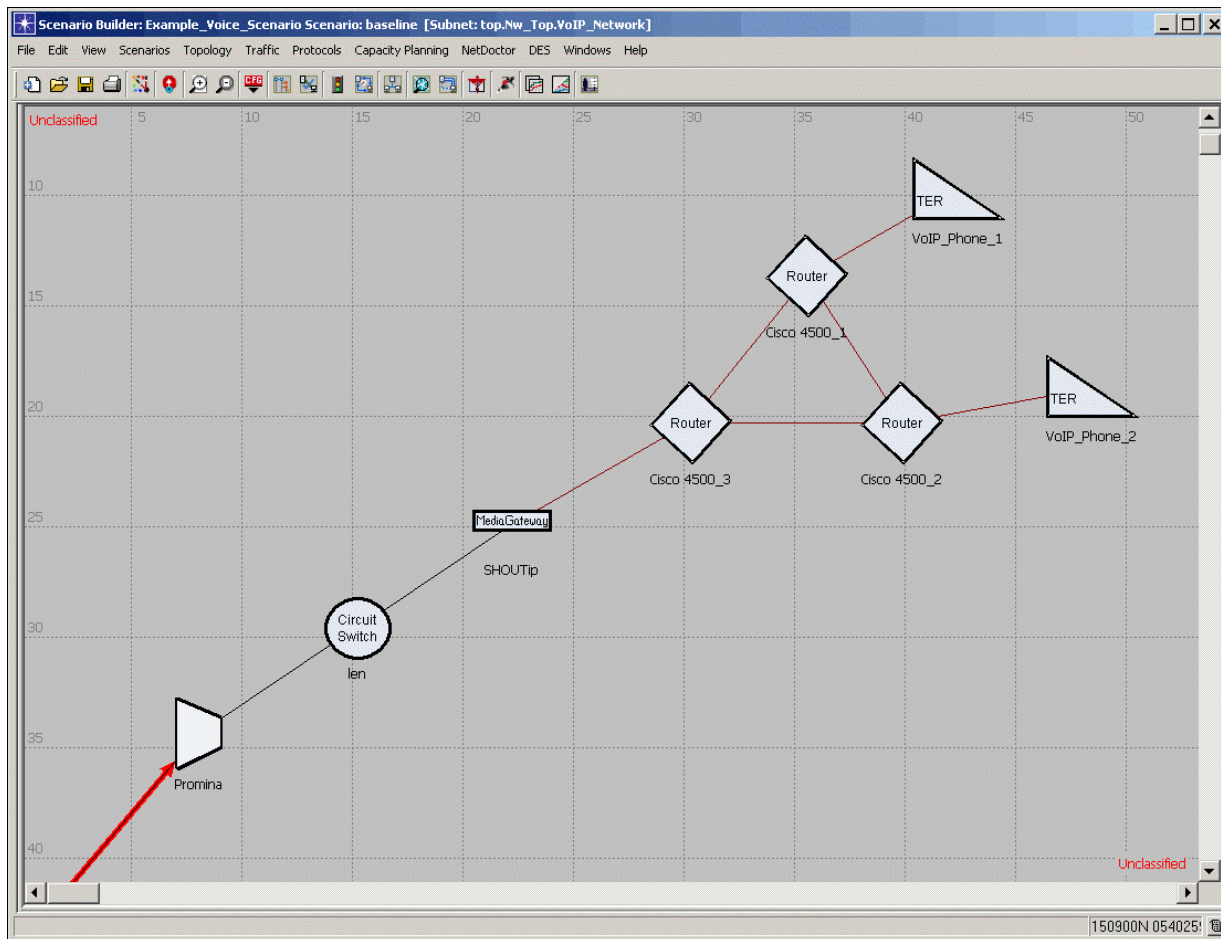*Figure 19: View of the Circuit_Switch_Network OPFAC*

Figure 20: View of the VoIP_Network OPFAC

3.1 Capacity Planning Reports

The Capacity Planner is a tool which allows you to quickly study a network by analytically calculating the performance. This provides you with a large amount of studies such as failure, configuration, and capacity studies. Inside NETWARS, the Capacity Planner tool has been enhanced to generate web reports. The web reports shown below are helpful when looking at the Voice devices. The numbers and statistics inside the web report examples are based on Capacity Planner runs using the example project shown above.

To run Capacity Planner on a network, click the **Capacity Planning > Evaluate…** menu after opening a particular NETWARS scenario. Refer to the NETWARS documentation for more information regarding how to use the Capacity Planner tool.

Once the Capacity Planner tool is run, the links in the network will be color-coded (to show utilization) and a web report will be shown displaying the results from the analysis. There are two parts of the web report which are specific to Voice devices (the other parts can be utilized but are not in the scope of this document). The first report is the **Executive Summary** portion of the report. This is the first page that displays after the

Capacity Planner tool has finished. This report discusses an overall view of the network and gives you high level points so that the user can easily narrow down problems areas.

Inside the Executive Summary is a section titled **Link Utilization**. This section shows the link with the maximum utilization out of all the links inside the scenario. This gives the user a quick insight on whether there are any bottlenecks (i.e., over utilized links) inside the network. Secondly, the section titled **Traffic Performance** is useful in making sure that all Traffic instances (IERs or Applications) have been routed successfully inside the network. A successful route of a traffic instance is based purely on the graph connectivity between the source and destination phones. Finally, the section titled **Circuit Performance** is useful in determining whether any circuit instances in the network were configured properly. In the case of Voice models, this section will not typically be used. However, in the example scenario above, a Promina circuit was created to bridge the Circuit Switch and IP networks.

It is important to note that the Capacity Planner will route traffic based on graph connectivity. In the case of the Voice models, the Capacity Planner will look at the voice channels available, IP connectivity, and whether circuits were deployed inside the network. The Capacity Planner will then route traffic based on the shortest path in the graph from the source to the destination. This means that certain protocols will not be used (such as IP, H.323, SIP, etc.) to route traffic inside Capacity Planner. For a more detailed analysis of voice calls with these protocols, the user should run the Discrete Event Simulation (DES). However, Capacity Planner will show the user several important metrics before investing time in running DES which can take much longer periods of time to finish. Please refer to the NETWARS documentation for more details on Capacity Planner.
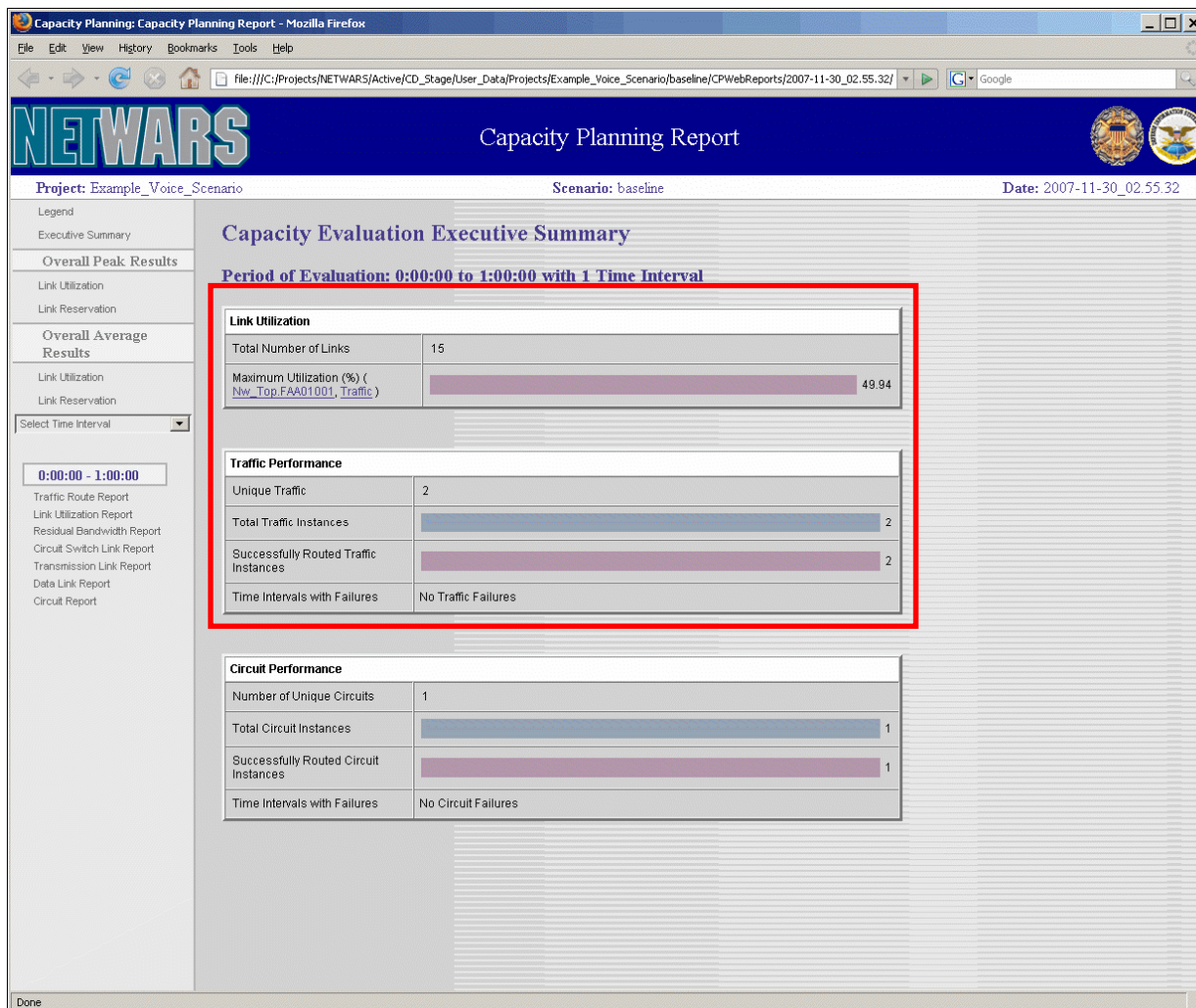
Figure 21: Capacity Planning Report Executive Summary

The **Link Utilization Report** is another section inside the Capacity Planning web report. This section can be accessed by clicking on the hyperlink located on the left side of the Capacity Planning report. The Link Utilization report shows a complete list of links inside the network. It provides a quick way of seeing all of the utilization numbers for each link inside the network. This allows you to easily debug any problems where traffic is could be providing a bottleneck to a particular link. This is also a continuation of the Executive Summary which only showed the link with the maximum utilization. Also, much of the data displayed is shown in hyperlink form so the user can click the necessary object in the report and the Scenario Builder will display the relevant link. The user can then check the necessary attributes or devices to look for configuration issues. Note that the rows in the report are color-coded to represent the utilization of the link (blue is zero, green is low, yellow is medium, and red is high utilization).
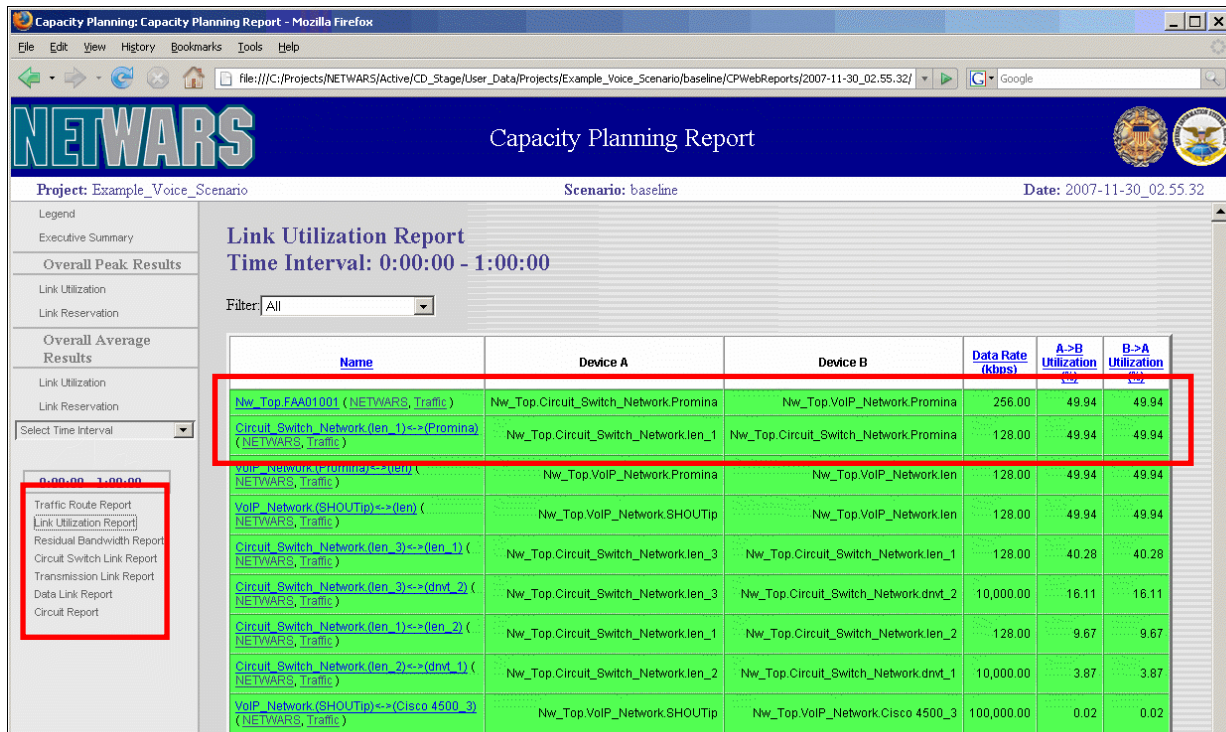
Figure 22: Capacity Planning Link Utilization Report

The **Traffic Route Report** is another section inside the Capacity Planning web report. Similar to the **Link Utilization Report**, this section can be accessed by clicking on the hyperlink located on the left side of the Capacity Planning report. The Traffic Route Report provides the ability to look at the different instances of traffic inside the scenario. This report is important for looking at where the traffic was being routed, how the traffic was defined, whether the traffic was routed, and what are the failure reasons. This can help the user determine whether the network has graph connectivity between the source and destination nodes. Again, this report is hyperlinked to help the user easily determine the configuration of the devices involved with the traffic instance.

Figure 23: Capacity Planning Traffic Route Report

The **Residual Bandwidth Report** is another section inside the Capacity Planning web report. Similar to the **Link Utilization Report**, this section can be accessed by clicking on the hyperlink located on the left side of the Capacity Planning report. The Residual Bandwidth Report provides the ability to look at the remaining bandwidth for both directions (forward and reverse) on a particular link. This report takes the total bandwidth of the link and subtracts the bandwidth out of the link which was used by the traffic defined in the scenario. This can help the user determine the amount of bandwidth available, and whether traffic should be added or subtracted from the link. The report can also assist in checking whether valid backup routes are available in the scenario. Again, this report is hyperlinked to help the user easily determine the devices and links involved in the report and to check the configuration of these objects. Note that the rows in the report are color-coded to represent the percentage of bandwidth used on the link (blue is zero, green is low, yellow is medium, and red is high).
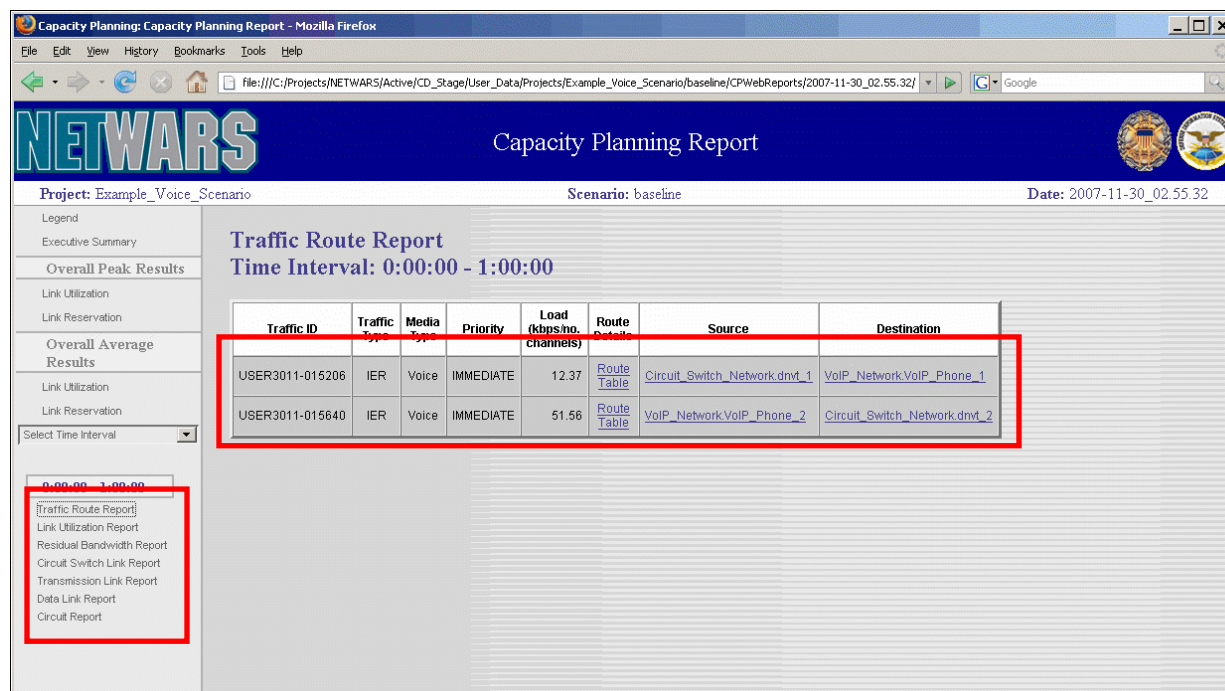
Figure 24: Capacity Planning Residual Bandwidth Report

Finally, the **Circuit Switch Link Report** is another section inside the Capacity Planning web report. Similar to the **Link Utilization Report**, this section can be accessed by clicking on the hyperlink located on the left side of the Capacity Planning report. The Circuit Switch Link Report provides the ability to look at the Circuit Switch links inside the scenario. This report allows the user to see the amount of utilization on the link, the average utilization on the voice channel, and the number and size of the voice channels on the link. This can help the user determine how well the circuit switch links and voice channels are utilized in the scenario. Again, this report is hyperlinked to help the user easily determine the devices and links involved in the report and to check the configuration of these objects. Note that the rows in the report are color-coded to represent the utilization of the link (blue is zero, green is low, yellow is medium, and red is high utilization).
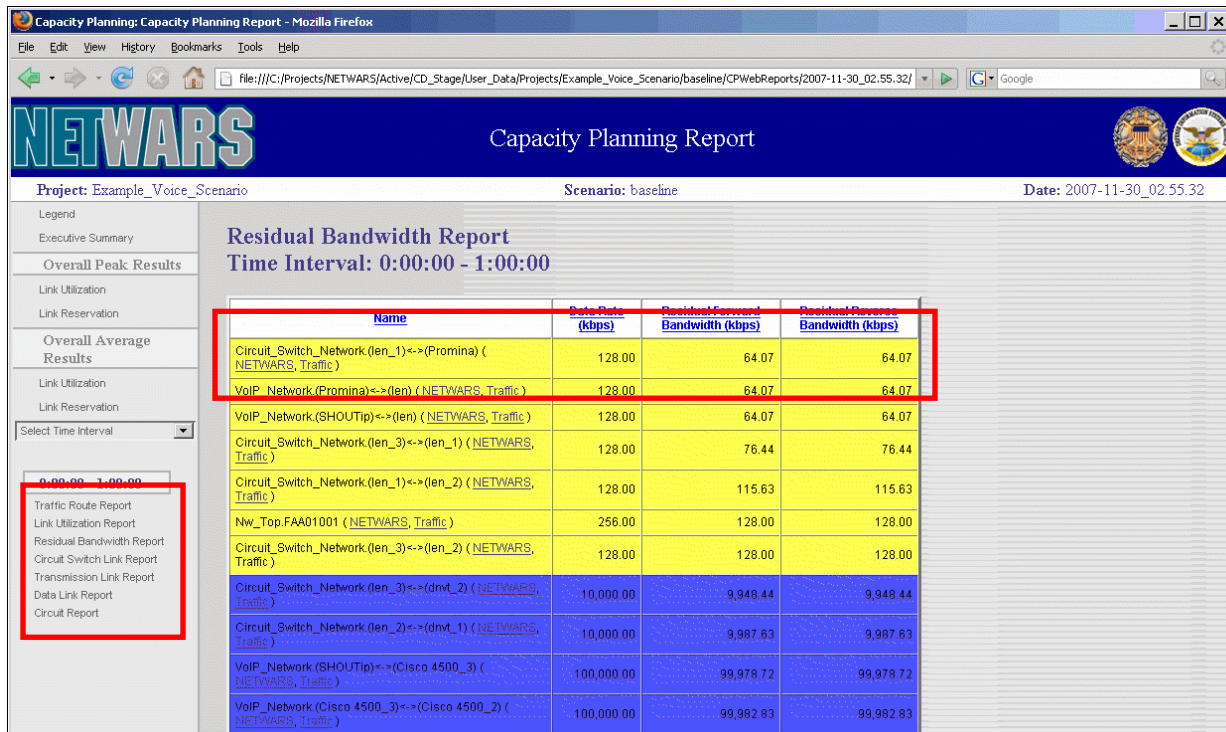
Figure 25: Capacity Planning Circuit Switch Link Report

3.2 Discrete Event Simulation Statistics

The Discrete Event Simulation (DES) allows you to simulate the performance of a network in detail by actually sending packets and data across a created NETWARS scenario. The statistics gathered during this simulation allow you to gain specific details about how a network and particular devices are performing. This information provides you with a widespread picture of the simulated network and allows you to perform a wide variety of studies. The listed statistics below are the specific statistics available when using the NETWARS Voice models. Many other DES statistics are available based on the type of technologies deployed inside the scenario. However, as this document is based purely on the Voice devices, the other available statistics are outside of the scope of this document. Refer to the NETWARS and OPNET IT Guru standard documentation for more details on other statistics.

Voice DES global statistics refer to statistics that are defined on a global network level basis. To define these type of statistics, right-click on the scenario workspace (not on a device, link, etc.) and select the **Choose Individual DES Statistics** menu item. After the **Choose Results** dialog box displays, navigate to the **Global Statistics** section. The global statistics from this section which work with the Voice models are shown below. Note that other global statistics are available but the listed statistics are used purely with the Voice models.

Table 12: DES Voice Global Statistics

Statistic	Description
H323.Setup Time	This statistic holds the average setup time (in seconds) for all H.323 calls in the network.
H323.Total Calls	This statistic holds the total number of H.323 calls signaled in the network during a simulation.
H323.Total Failed Calls	This statistic holds the total number of H.323 calls in the network which had a signaling failure during simulation.
H323.Total Successful Calls	This statistic holds the total number of H.323 calls in the network which were signaled successfully during a simulation.
SIP.Call Setup Time (sec)	This statistic holds the average setup time (in seconds) for all SIP calls in the network.
SIP.Calls Rejected	This statistic holds the total number of rejected SIP calls in the network.
Total Voice Traffic.Blocking Probability	This statistic specifies the probability that a Voice IER was blocked in the network. A blocked IER means that no device was available to accept the particular IER traffic at the start time and it must be retried at a later time.
Total Voice Traffic.Call Completion Rate	This statistic specifies the probability that a Voice call is completed on the network. A completed Voice call means that the initial call setup occurred and all data was sent for the duration of the call.
Total Voice Traffic.Connection Latency	This statistic specifies the average amount of time (in seconds) that a Voice call requires to initially setup in the network.
Total Voice Traffic.End to End Delay	This statistic specifies the average amount of delay (in seconds) for a Voice packet to reach the destination in the network. This is based off the sent time of the IER (i.e., when the IER was sent to the device).
Total Voice Traffic.Grade of Service	This statistic specifies the percentage that a Voice call was able to be sent onto the network (i.e., it did not perish).
Total Voice Traffic.IER Failed Count	This statistic specifies the total number of failed Voice calls (IERs) in the network. Failed Voice calls are calls which could not be setup or complete inside the network.
Total Voice Traffic.IER Perished Count	This statistic specifies the total number of perished Voice calls (IERs) in the network. A perished Voice call occurs when the IER is continually blocked until the specified perish time has been reached for the IER. This means that the IER was unable to be sent in a reasonable amount of time (as specified by the IER definition).
Total Voice Traffic.IER Sent Count	This statistic specifies the total number of sent Voice calls (IERs) in the network. A sent call occurs when the IER is started on a particular device (i.e., it was not blocked) and setup begins.
Total Voice Traffic.Message Completion	This statistic specifies the total number of completed Voice calls (IERs) in the network. A completed Voice call means

	that the initial call setup occurred and all data was sent for the duration of the call.
Total Voice Traffic.Speed of Service	This statistic specifies the average amount of delay (in seconds) for a Voice packet to reach the destination in the network. This is based off the start time of the IER (i.e., when the call signaling was started on the device).
Total VTC Traffic.Blocking Probability	This statistic specifies the probability that a VTC IER was blocked in the network. A blocked IER means that no device was available to accept the particular IER traffic at the start time and it must be retried at a later time.
Total VTC Traffic.Call Completion Rate	This statistic specifies the probability that a VTC call is completed on the network. A completed VTC call means that the initial call setup occurred and all data was sent for the duration of the call.
Total VTC Traffic.Connection Latency	This statistic specifies the average amount of time (in seconds) that a VTC call requires to initially setup in the network.
Total VTC Traffic.End to End Delay	This statistic specifies the average amount of delay (in seconds) for a VTC packet to reach the destination in the network. This is based off the sent time of the IER (i.e., when the IER was sent to the device).
Total VTC Traffic.Grade of Service	This statistic specifies the percentage that a VTC call was able to be sent onto the network (i.e., it did not perish).
Total VTC Traffic.IER Failed Count	This statistic specifies the total number of failed VTC calls (IERs) in the network. Failed VTC calls are calls which could not be setup or complete inside the network.
Total VTC Traffic.IER Perished Count	This statistic specifies the total number of perished VTC calls (IERs) in the network. A perished VTC call occurs when the IER is continually blocked until the specified perish time has been reached for the IER. This means that the IER was unable to be sent in a reasonable amount of time (as specified by the IER definition).
Total VTC Traffic.IER Sent Count	This statistic specifies the total number of sent VTC calls (IERs) in the network. A sent call occurs when the IER is started on a particular device (i.e., it was not blocked) and setup begins.
Total VTC Traffic.Message Completion	This statistic specifies the total number of completed VTC calls (IERs) in the network. A completed VTC call means that the initial call setup occurred and all data was sent for the duration of the call.
Total VTC Traffic.Speed of Service	This statistic specifies the average amount of delay (in seconds) for a VTC packet to reach the destination in the network. This is based off the start time of the IER (i.e., when the call signaling was started on the device).
Voice.Jitter (sec)	This statistic specifies the average difference in latency (in seconds) between two packets on the network.
Voice.Packet Delay Variation	This statistic holds the variance among end-to-end delays for voice packets. The end-to-end delay for a voice packet is measured from the time it is created to the time it is received.
Voice.Packet End-to-End Delay (sec)	This statistic specifies the average amount of delay (in seconds) for a Voice packet to reach the destination in the network.
Voice.Traffic Received (bytes/sec)	This statistic holds the average number of bytes per second forwarded to all Voice applications by the transport layers in the network.
Voice.Traffic Received (packets/sec)	This statistic holds the average number of packets per second

	forwarded to all Voice applications by the transport layers in the network.
Voice.Traffic Sent (bytes/sec)	This statistic holds the average number of bytes per second submitted to the transport layers by all voice applications in the network.
Voice.Traffic Sent (packets/sec)	This statistic holds the average number of packets per second submitted to the transport layers by all voice applications in the network.

Voice DES node statistics refer to statistics that are defined at the node level. To define these type of statistics, right-click on the scenario workspace (not on a device, link, etc.) and select the **Choose Individual DES Statistics** menu item. After the **Choose Results** dialog box displays, navigate to the **Node Statistics** section. The selected statistics from this section will then be collected for every device in the network which can write this particular statistic. If the user wishes to specify statistics on a per-node basis instead, the user should right-click directly on the device select **Choose Individual DES Statistics**. The node statistics from this section which work with the Voice models are shown below. Note that other node statistics are available but the listed statistics are used purely with the Voice models.

Table 13: DES Voice Node Statistics

Statistic	Description
H323.Active Calls	This statistic holds the number of active calls on this H.323 device at a given time period.
H323.Setup Time	This statistic holds the average setup time (in seconds) for a H.323 call on this node.
H323.Total Calls	This statistic holds the total number of H.323 calls signaled on this node during a simulation.
H323.Total Failed Calls	This statistic holds the total number of H.323 calls on this node which had a signaling failure during simulation.
H323.Total Successful Calls	This statistic holds the total number of H.323 calls on this node which were signaled successfully during a simulation.
SHOUTip.Active Calls	This statistic holds the number of active calls on this gateway at a given time period.
SHOUTip.Total Calls	This statistic holds the total number of calls placed on this gateway during a simulation.
SHOUTip.Total Successful Calls	This statistic holds the total number of failed calls on this gateway during a simulation.
SHOUTip.Total Failed Calls	This statistic holds the total number of successful calls on this gateway during a simulation.
SHOUTip.Jitter (sec)	This statistic holds the jitter (in seconds) between the previously received packet and the currently received packet. Jitter refers to the difference in latency between two packets.
SHOUTip.Packet Delay (sec)	This statistic holds the average delay (in seconds) for each packet received by this device.
SHOUTip.Total Data Sent Over IP Network	This statistic holds the total amount of data sent over the IP network by this gateway during a simulation.
SIP UAC.Active Calls	This statistic holds the number of SIP calls on this device at a given time period.
SIP UAC.Call Duration (sec)	This statistic specifies the average duration (in seconds) of

	each SIP call on this device. Duration is defined as the time which the node got call connect confirmation to the time at which it got call disconnect confirmation.
SIP UAC.Call Setup Time (sec)	This statistic specifies the average time (in seconds) needed to setup the SIP call.
SIP UAC.Calls Connected	This statistic specifies the total number of SIP calls connected. This includes both calls initiated by this node and any incoming call requests.
SIP UAC.Calls Initiated	This statistic specifies the total number of SIP calls initiated at his node.
SIP UAC.Calls Rejected	This statistic specifies the total number of SIP calls rejected on this device. This includes both calls initiated by this node and any incoming call requests.
VoIP Phone.Active Calls	This statistic holds the number of active calls on this phone at a given time period.
VoIP Phone.Total Calls	This statistic holds the total number of calls placed on this phone during a simulation.
VoIP Phone.Successful Calls	This statistic holds the total number of successful calls on this phone during a simulation.
VoIP Phone.Failed Calls	This statistic holds the total number of failed calls on this phone during a simulation.
VoIP Phone.Call Setup Time	This statistic holds the average time (in seconds) that calls traversing this node take to setup properly (i.e., perform the initial signaling).
VoIP Phone.Jitter (sec)	Statistic holds the jitter (in seconds) between the previously received packet and the currently received packet. Only available on the H.323 terminals.
VoIP Phone.Generated Phone Calls	Statistic holds the number of calls placed on this phone where this phone is acting as the source (caller) phone. Only available on the H.323 terminals.
VoIP Phone.Received Phone Calls	Statistic holds the number of calls placed on this phone where this phone is acting as the destination (callee) phone. Only available on the H.323 terminals.
VoIP Phone.Packet Delay (sec)	Statistic holds the average delay (in seconds) for each packet received by this device. Only available on the H.323 terminals.
VoIP Phone.Total Data Sent Over IP Network	Statistic holds the total amount of data sent over the IP network by this phone during a simulation. Only available on the H.323 terminals.

Voice DES link statistics refer to statistics that are defined at the link level. To define these type of statistics, right-click on the scenario workspace (not on a device, link, etc.) and select the **Choose Individual DES Statistics** menu item. After the **Choose Results** dialog box displays, navigate to the **Link Statistics** section. The selected statistics from this section will then be collected for every link in the network which can write this particular statistic. If the user wishes to specify statistics on a per-link basis instead, the user should right-click directly on the link and select **Choose Individual DES Statistics**. The link statistics from this section which work with the Voice models are shown below. Note that other link statistics are available but the listed statistics are used purely with the Voice models.

Table 14: DES Voice Link Statistics

Statistic	Description
Voice Channels.Channel Utilization	This statistic specifies the utilization for a Circuit Switch voice channel inside the scenario. This is calculated by the voice call bit rate divided by voice channel size.

3.3 DES Reporting Capabilities

In addition to the Discrete Event Simulation (DES) statistics, extra DES reporting capabilities can be added to help the user obtains results. These reports allow the user to access information which would otherwise be difficult to show statistically. Also, these reports give the user an easy way to navigate through a large amount of information. As part of the H.323 model suite, these reporting capabilities were developed and can be accessed through the **DES > Results > View Results…** menu (after a Discrete Event Simulation has been run). Please refer to the NETWARS H.323 Model User Guide for further details.

3.4 Logical Views

Logical Views provide the ability to filter the network so that the user can look at particular layers or technologies more closely. In the case of Voice devices, the user should use the **IP** and **Voice** layers. This layer can be located under the **Network Layer Filters** section of the logical links dialog box. Secondly, the user can utilize the **Complete Network** layer with Voice devices. Note that the logical links functionality can be found under the **View > Show Logical Views…** menu (dialog box shown below).
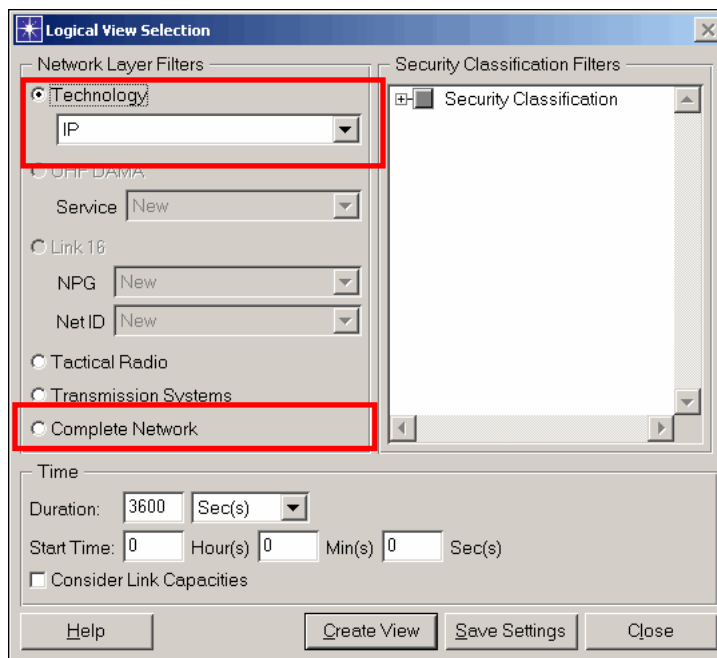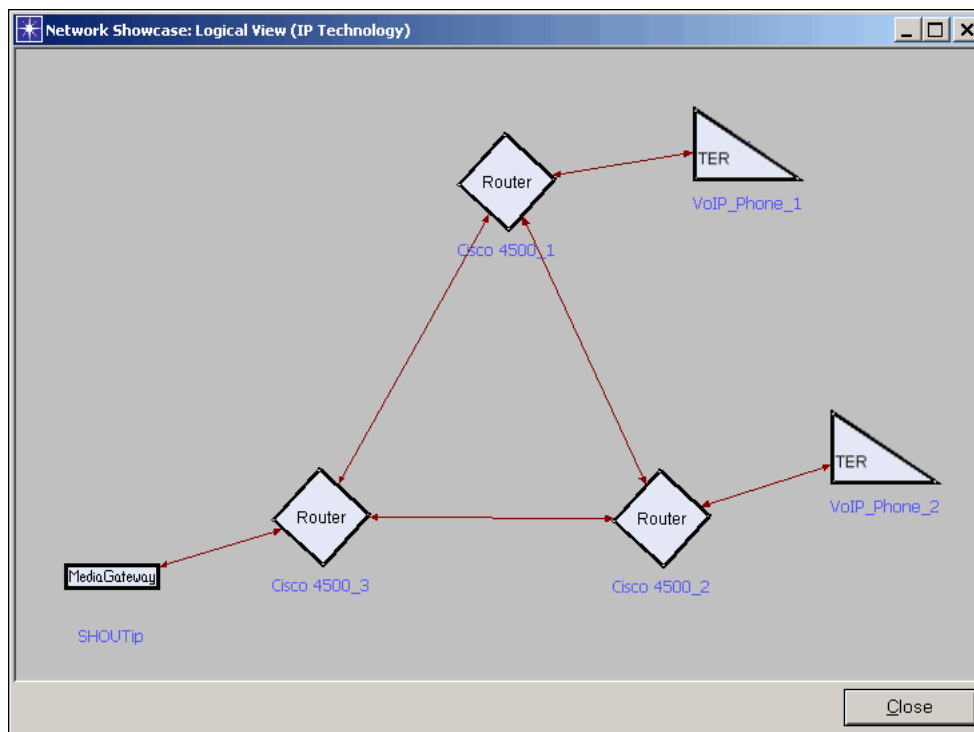

Figure 26: Logical View Dialog Box

Once an **IP** logical view of the Voice devices is created, the user is able to see a graphical representation of the IP connectivity of the Voice devices inside the scenario. The figure provided below shows the IP connectivity of the example project above. In the case of Voice devices, the user should expect to see the connectivity between the VoIP technology devices. Using the example project, the user will see that most of the devices located inside the "VoIP_Network" OPFAC are displayed below.

Figure 27: IP Technology Logical View Example

Once a **Voice** logical view of the Voice devices is created, the user is able to see a graphical representation of the Circuit Switch connectivity of the Voice devices inside the scenario. The figure provided below shows the Circuit Switch connectivity of the example project above. Using the example project, the user will see that most of the devices located inside the "Circuit_Switch_Network" OPFAC are displayed below.

It is important to note the orange, logical link which is displayed between the two **LEN** devices. In the example case, this logical link is abstracting the layer-2 Promina connection which sits between these two devices. What this link shows to the user is that these two **LEN** devices can "talk" to each other using the Circuit Switch technology even though they are not physically connected to each other. To find more information about the Logical Link, right-click on the link and select **Edit Attributes**. This will show information such as the physical links which make up the connection, the date rate, etc. Note that Logical Links can occur in the other Logical View filters to abstract technologies which are not relevant to the filter technology.
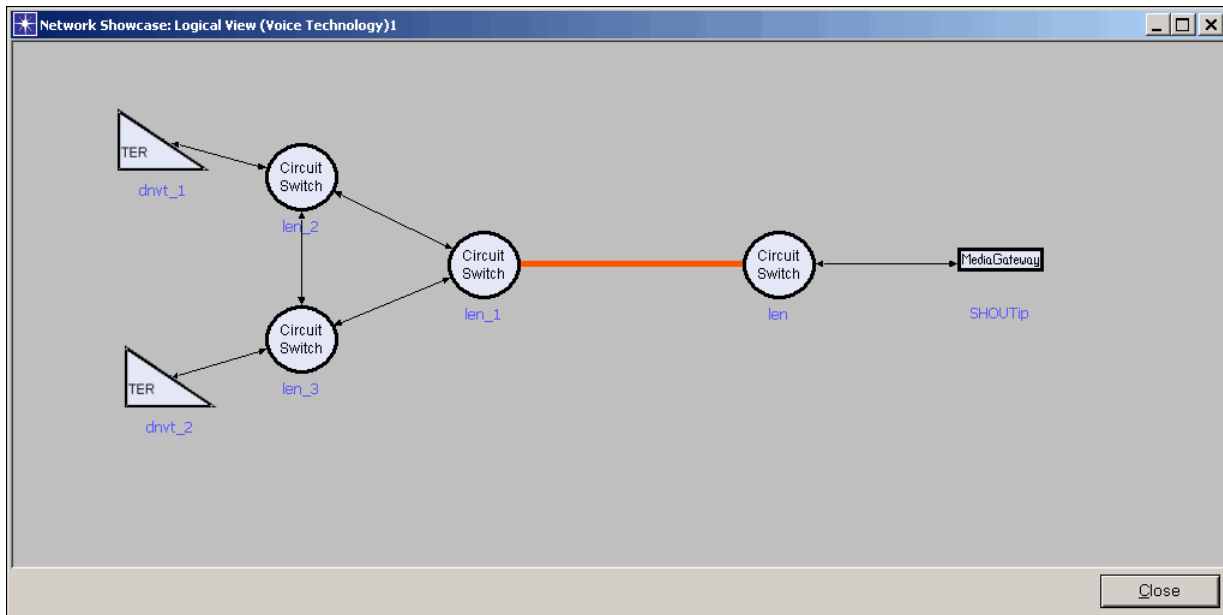
Figure 28: Voice Technology Logical View Example

Once a **Complete Network** logical view of the Voice devices is created, the user is able to see a graphical representation of the entire network without OPFACs or Organizations. Essentially, this is the network shown with physical connections only. The figure provided below shows the network connectivity of the example project above. Note that the red link shown between the Promina devices is the Promina circuit object.
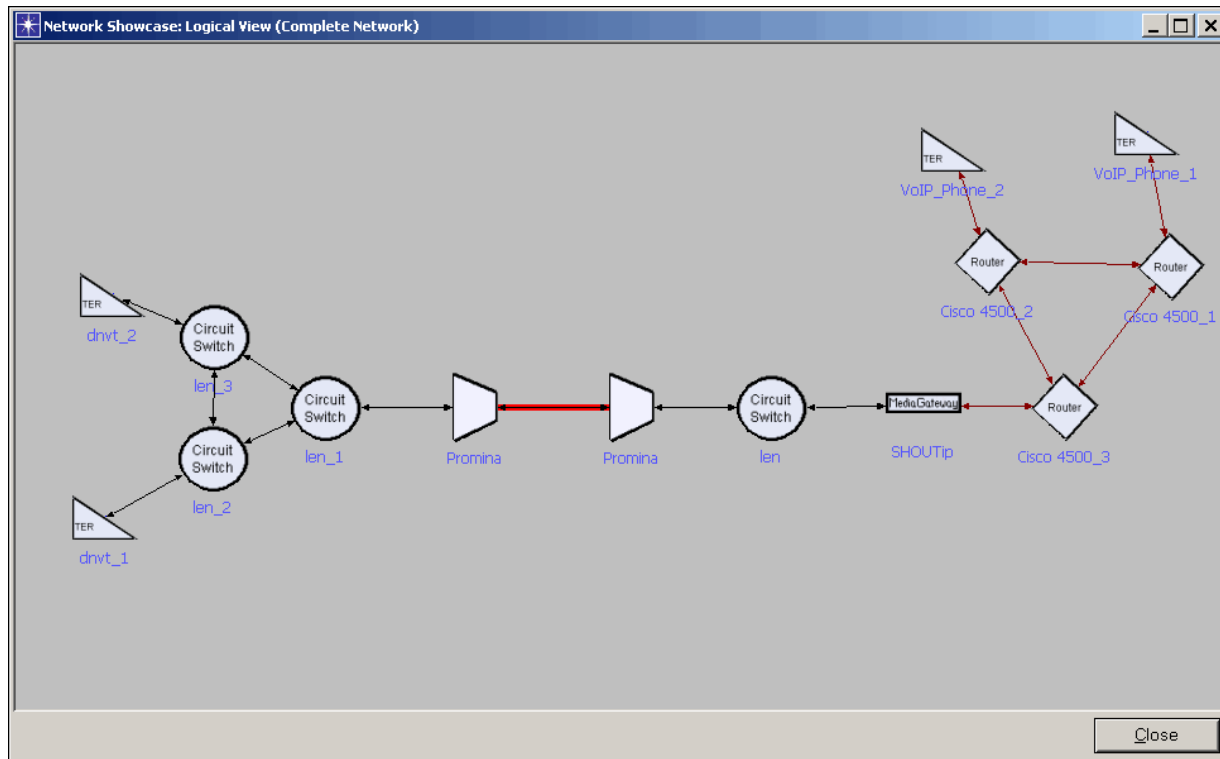
Figure 29: Complete Network Logical View Example

To view more information on the Logical Views functionality, refer to the *NETWARS User's Manual* for more details.

3.5   Example Equipment Strings

Equipment strings are combinations of connected devices which can typically be found inside a network. The below equipment strings should be considered as normal use cases for the Voice devices and are supported by NETWARS. Note that arrows denote links inside the network.

This document does not show the equipment strings related to H.323 and SIP setups. For exact specifications on how to setup these types of networks, consult the H.323 or SIP documentation.

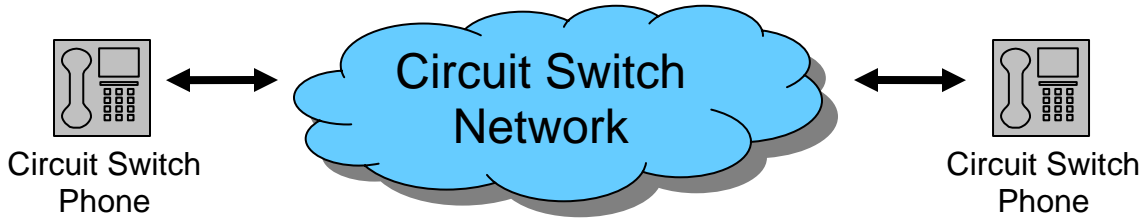Any additional equipment strings not shown could still be supported by the Voice models.

Figure 30: Circuit Switch Phone to Circuit Switch Phone Equipment String
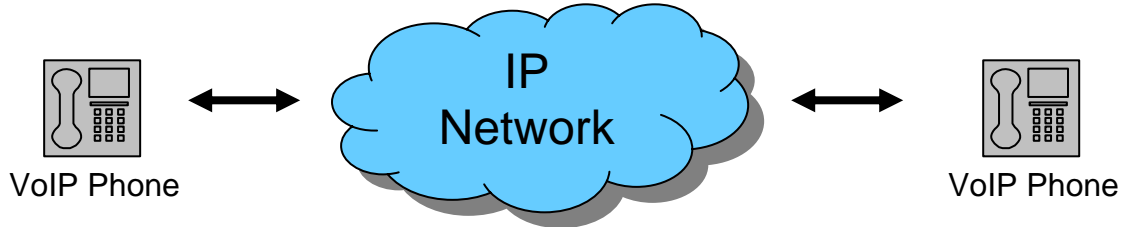

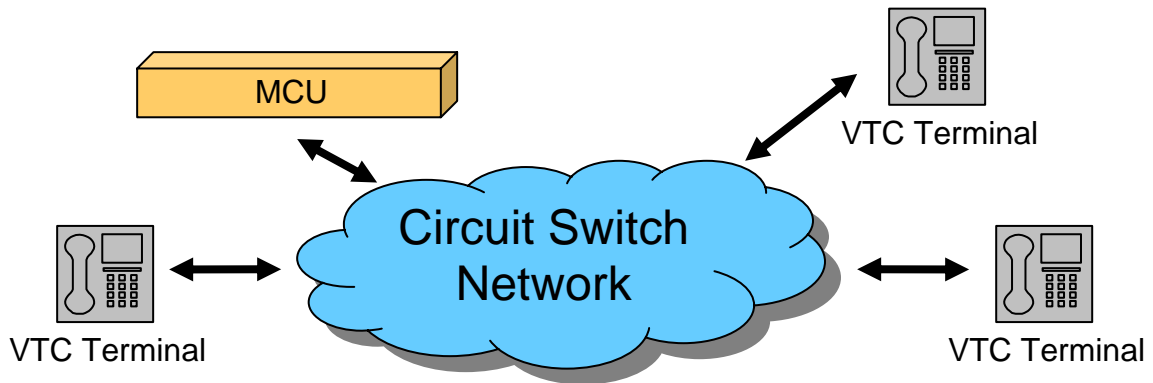*Figure 31: VoIP Phone to VoIP Phone Equipment String*
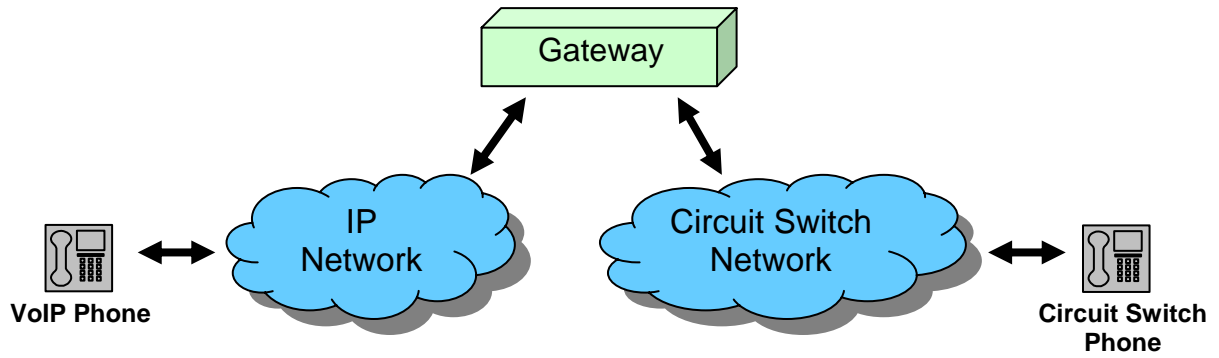

Figure 32: VTC Equipment String

Figure 33: VoIP to Circuit Switch Equipment String
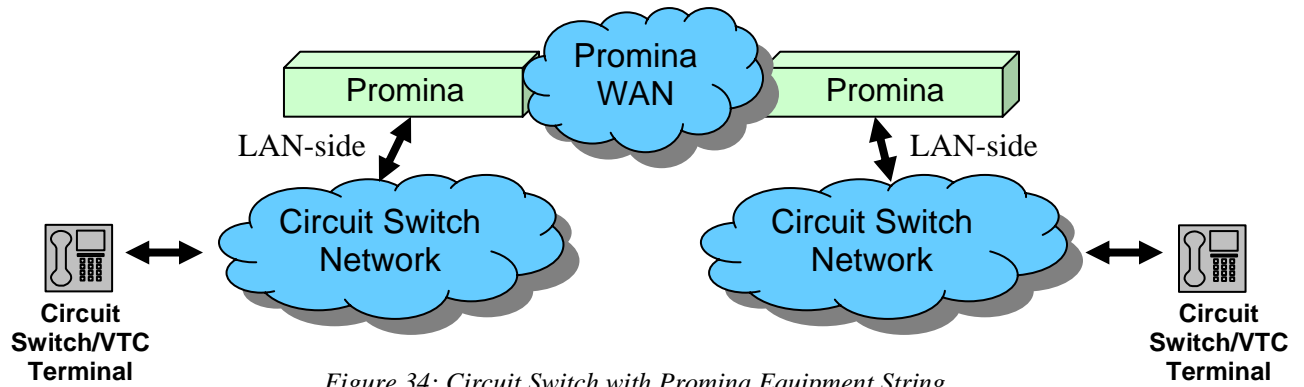

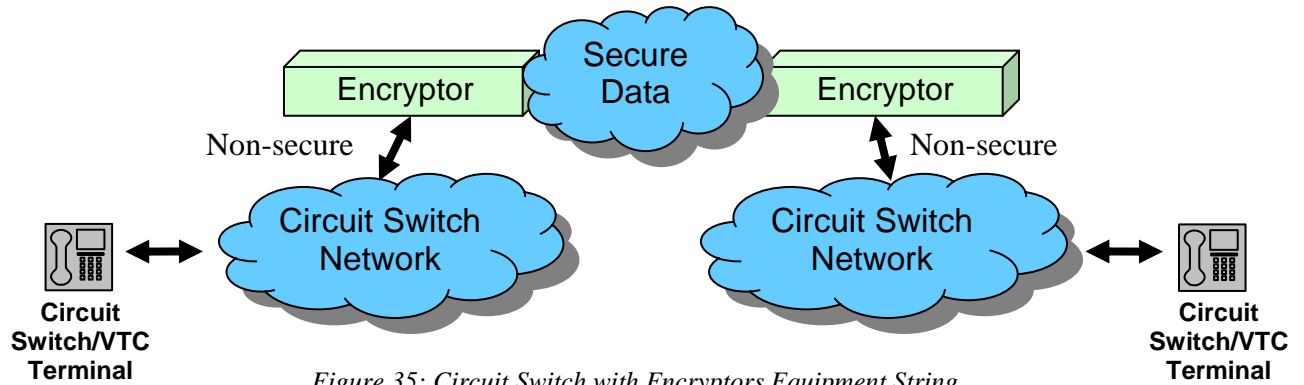*Figure 34: Circuit Switch with Promina Equipment String*

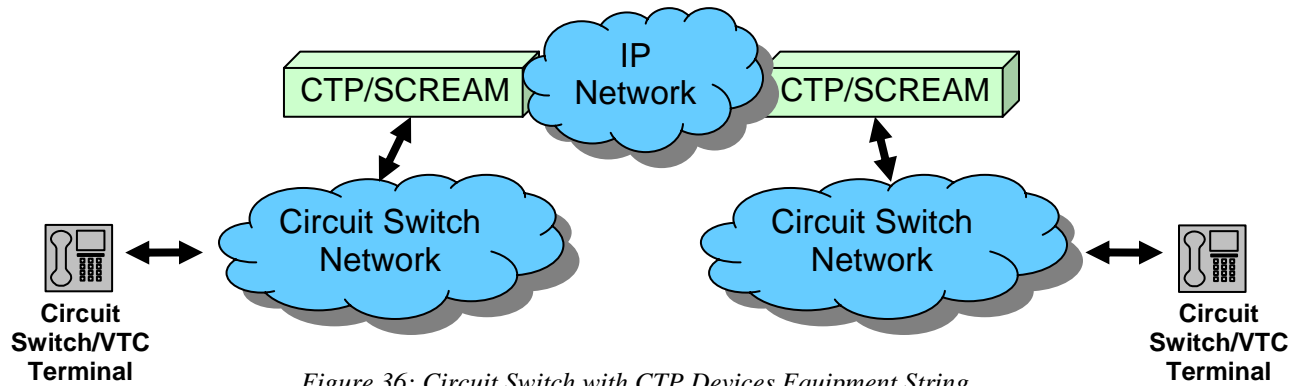*Figure 35: Circuit Switch with Encryptors Equipment String*

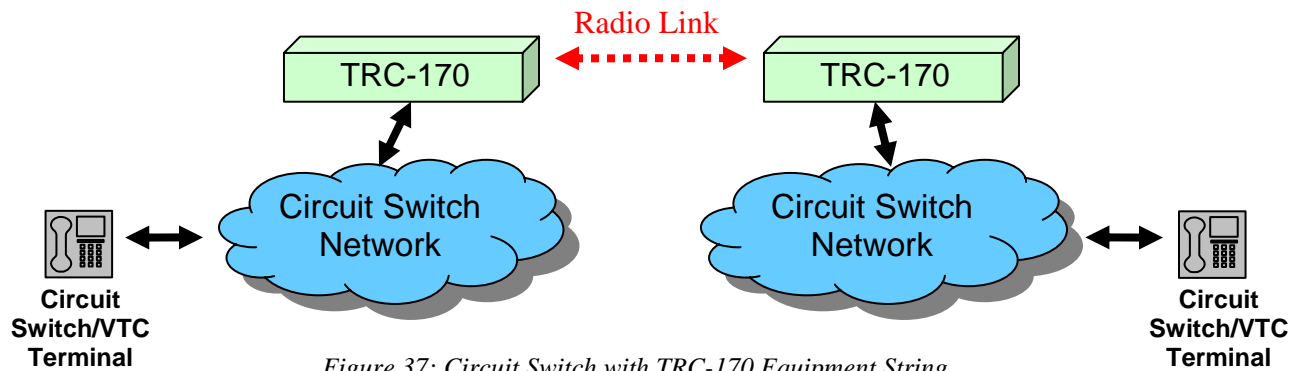*Figure 36: Circuit Switch with CTP Devices Equipment String*

Figure 37: Circuit Switch with TRC-170 Equipment String

3.6   Traffic Usage

Depending on the Voice technology, different traffic types are available. The Circuit Switch and VTC models work strictly with NETWARS IER traffic. The VoIP models work with both NETWARS IERs and standard OPNET traffic (Applications, Flows, ACE, etc.).

3.7   Usage in Modeler

Currently, the NETWARS Circuit Switch and VTC models cannot easily be imported into OPNET Modeler. This is due to the heavy reliance upon the NETWARS IER traffic. For these types of models, the user would need to remove all of the NETWARS IER references inside the model code and replace the IER code with a standard OPNET traffic. Once this step is performed, the user could then import the models into OPNET Modeler successfully by adding them into the **mod_dirs** model directories (accessible in the **Edit > Preferences** menu).

For the VoIP models, these models can easily be imported into OPNET Modeler. Please refer to the *H.323 Models User's Guide* for information on how to import these models.

4 Node Models

4.1 Architecture

The figure below shows the node models for the Voice devices. These node models provide an example of the full range of features available in the NETWARS Voice model suite. The specific process models which make up the modules inside the node models are out of scope for this document. However, all of the Voice related devices and files can be located under the **<NETWARS Installation Directory>\Sim_Domain\op_models\netwars_std_models \circuit_switch_voice** and **<NETWARS Installation Directory>\Sim_Domain\op_models\netwars_std_models \voip** directories and sub-directories (where **<NETWARS Installation Directory>** is the directory where NETWARS is installed). Note that all of the files and models in this directory can be opened and modified using the OPNET Modeler software.

4.1.1 Circuit Switch Phone and VTC Node Models

The Circuit Switch Phone and VTC (VTC_Terminal and MCU) nodes represent the endpoints where all calls can be initiated and received (i.e., the caller and callee). As these terminals both work inside the Circuit Switch domain and are end stations, the node models between these two devices are similar.

To perform the designated functionality, a "se" module is added on top of Circuit Switch interfaces to create the complete node design. This application layer module will provide the Circuit Switch Voice functionality by allowing the user to specify voice calls between end terminals. All voice IERs sent to these terminals from the overlying "OE" node will be received by the "se" module and translated into actual Voice traffic by the *cs_voice_se* process. This functionality includes using the Flood Search Routing protocol for call setup and signaling, as well as, reserving the necessary bandwidth on the Phone Switch link connecting to the border Circuit Switch device.

Note that OPNET traffic (i.e., Applications and Flows) are not supported on these end devices.

A single Phone Switch interface has been added to give the terminals access to the Circuit Switch network. The Phone Switch interface is a special interface which is used only by end stations inside the Circuit Switch network. It essentially stands for the link which connects the phone and the first border circuit switch device (i.e., a circuit switch). This port can be recognized by the "pr_#_se_transmitter" name given to the transmitter module and the "pr_#_se_receiver" name given to the receiver module.
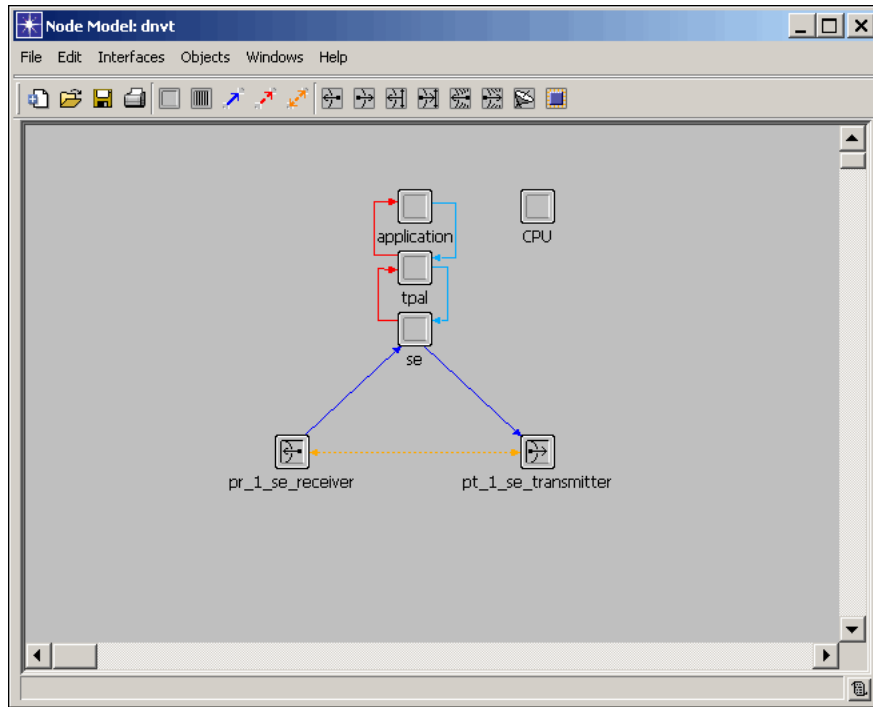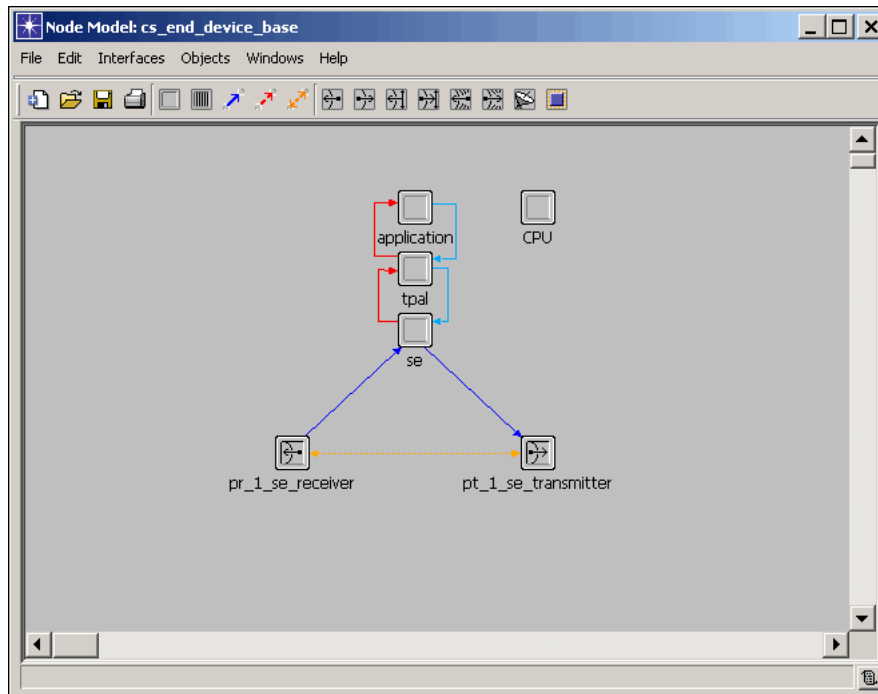
Figure 38: Circuit Switch Phone Node Model

Figure 39: VTC Terminal and MCU Node Model

4.1.2 Circuit Switch Devices Node Models

The Circuit Switch Device nodes (i.e., **LEN**, **SEN**, **TTC-39**, etc.) represent the circuit switches inside the Circuit Switch network. These switches forward the necessary signaling and data across the Circuit Switch network and reserve bandwidth on the links connected to them. As these devices all accomplish the same tasks, their node models are similar with the main difference being the number of ports each device contains. An example **LEN** node model is shown below.

To perform the designated functionality, "mux" and "circuit switch" modules are added next to the Phone Switch and Circuit Switch interfaces to create the complete node design. The "mux" module (which includes the *mux* process) is responsible for sending the packets received from the Phone Switch interfaces and passing them to the "circuit switch" module. It is also responsible for sending hello packets out on the Circuit Switch interfaces on regular intervals. Hello packets allow the different switches to check for network failure. If a hello packet is not received by a neighboring device within a certain time frame, that neighboring device and route is marked as failed. All calls which would use that particular route are also failed.

The "circuit switch" module (including the *circuit_switch* and *flood_search_routing* processes) is responsible for most of the Circuit Switch device behavior. The module performs all of the dynamic Flood Search Routing protocol signaling, as well as, reserving the bandwidth on the connected Circuit Switch links.

Several Phone Switch interfaces have been added to allow the Circuit Switch devices the ability to connect to Circuit Switch or VTC end stations. The Phone Switch interface is a special interface which is used only by end stations inside the Circuit Switch network. It essentially stands for the link which connects the phone and the Circuit Switch device. This port can be recognized by the "dtg_pt_#" name given to the transmitter module and the "dtg_pr_#" name given to the receiver module.

Several Circuit Switch interfaces have been added to allow the Circuit Switch devices to connect to each other, as well as, other types of devices which can support Circuit Switch (such as the Voice Gateway, Promina, Encryptors, TRC-170 radios, etc.). This port can be recognized by the "ckt_pt_#" name given to the transmitter module and the "ckt_pr_#" name given to the receiver module.
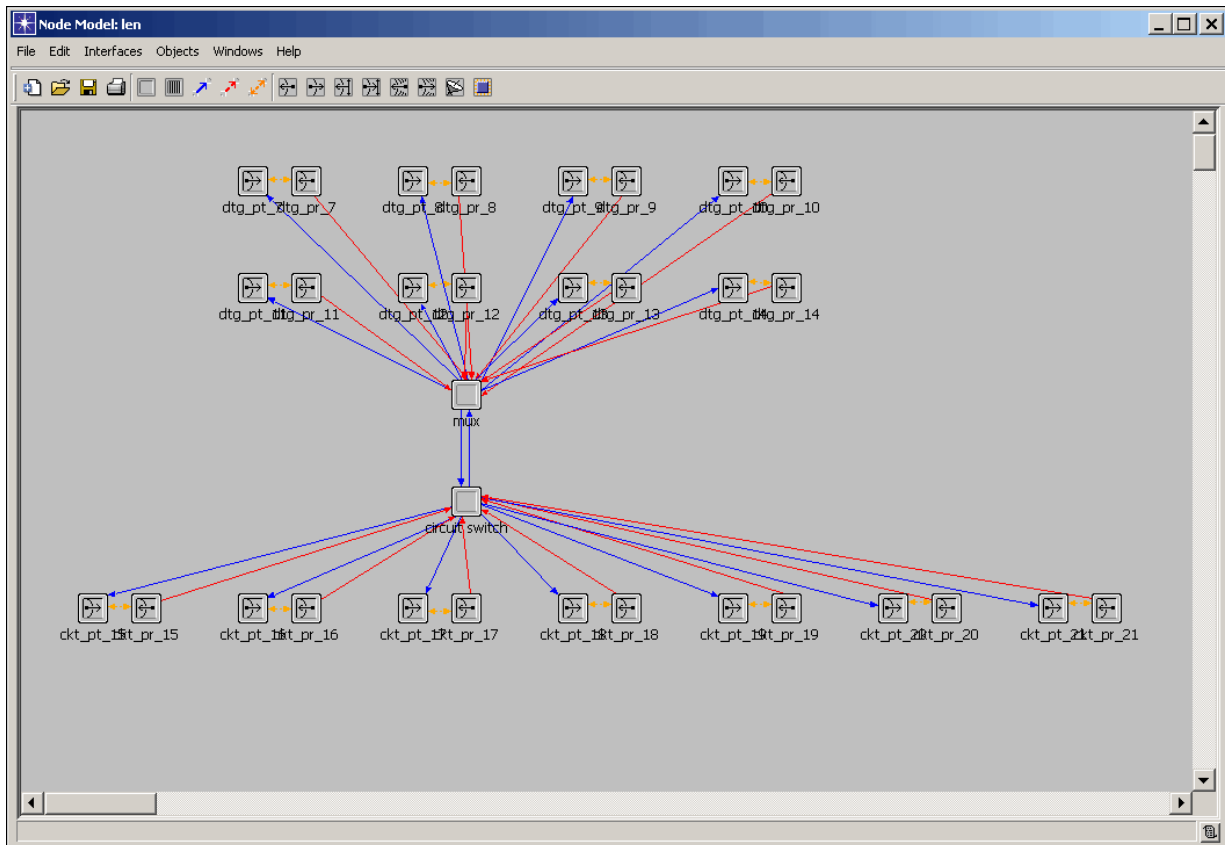
Figure 40: Circuit Switch Device Node Model

4.1.3 VoIP Phone Node Model

The VoIP Phone node represents the endpoint where all calls can be initiated and received (i.e., the caller and callee). Currently, this phone supports both SIP and H.323 signaling and allows the user to toggle between these different signaling protocols using the "Phone VoIP Protocol" attribute. The SIP protocol for this phone will remain as the default but H.323 will now be available.

To perform the designated functionality, both a "se" and "application" module are added to the standard OPNET IP stack to create the complete VoIP Phone node. These two application layer modules will provide the call signaling (H.323 and SIP) functionality by allowing the user to specify voice calls between end terminals. Also, these modules are responsible for creating the necessary data packets which will load the network for the duration of the call.

The "se" module is responsible for interacting with the standard NETWARS IER structure. All voice IERs sent to the VoIP Phone from the overlying "OE" node will be received by the "se" module and translated into actual voice traffic. This translation is performed by the *voip_se* process which can provide H.323 or SIP functionality by interfacing with its H.323 or SIP child processes.

The "application" module (including the *gna_clsvr_mgr* process) is provided to interface with the standard OPNET application framework. Support currently exists for VoIP applications inside the standard OPNET library which can use SIP signaling only (H.323 support will be added in OPNET version 14.5). This workflow allows the "application" module to translate any application definitions into realistic voice traffic. This is performed by the *gna_clsvr_mgr* process which interfaces with the *h323_mgr* and *sip_UAC* child process models (similar to the "se" module) to perform the necessary VoIP Phone signaling and functionality.

A single 10/100Mbps Ethernet interface has been added to the standard NETWARS IP stack to give the VoIP Phone access to the IP network. This port can be recognized by the "eth_tx_#_#" name given to the transmitter module and the "eth_rx_#_#" name given to the receiver module.
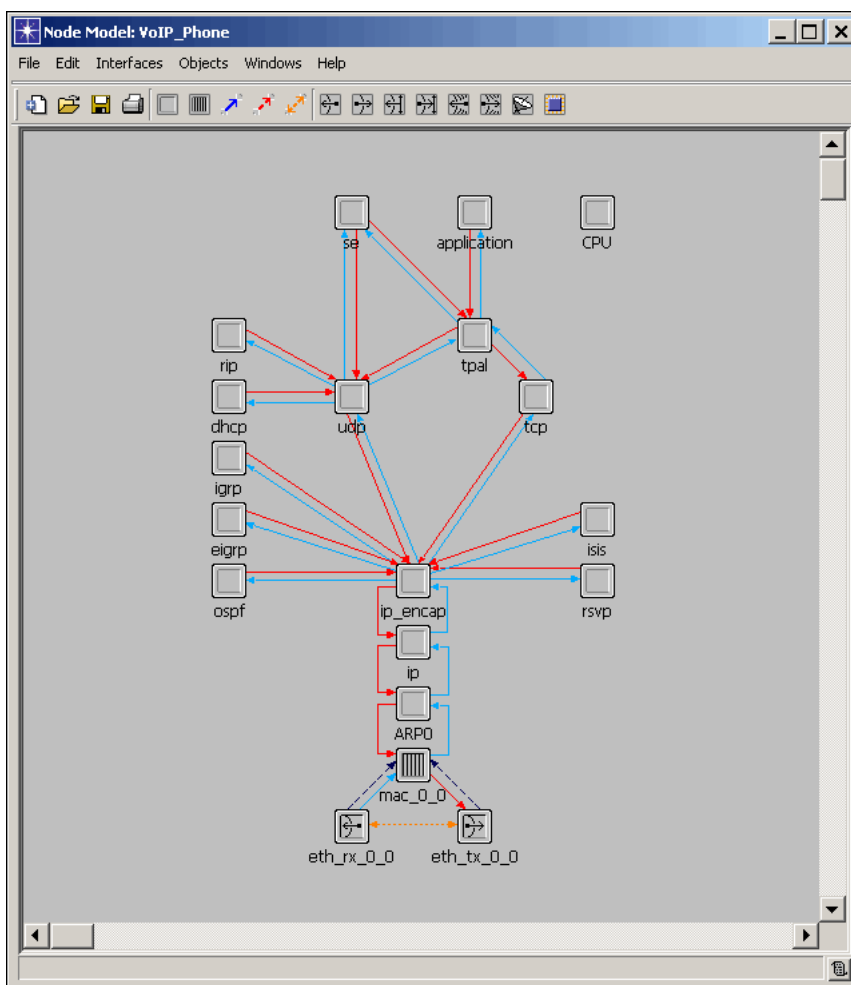

Figure 41: VoIP Phone Node Model

4.1.4 Voice Gateway Node Model

The Voice Gateway provides the ability to convert VoIP Voice calls into non-VoIP standard voice calls (and vice versa). This includes SIP to Circuit Switch, H.323 to Circuit Switch, Circuit Switch to SIP, Circuit Switch to H.323, and H.323 to SIP conversions. The Voice Gateway design will use the Media Gateway and SHOUTip devices (NETWARS 2007 enhancements) as the base models. Note that the Gateway can only interface with the NETWARS circuit switch devices and IER traffic. OPNET applications will not be supported through the Gateway as NETWARS Circuit Switch devices currently do not support OPNET applications.

The Media Gateway and N.E.T. SHOUTip devices contain many different types of functionality including VoIP (SIP and H.323) and SAToP support. Due to scope, this document will only reference the VoIP portions of these devices. Essentially, the VoIP portion of the Media Gateway and SHOUTip devices include the *nw_voice_dispatch* process. This process provides the necessary SIP/H.323 functionality by interfacing with the *h323_mgr*, *sip_UAC*, and *sip_UAS* child processes. Special care must be taken by the Voice Gateway as it can act as both a client (i.e., terminal/endpoint for Circuit Switch phones) for a call and as an H.323 Gatekeeper and/or SIP Proxy. Because of this, the *nw_voice_dispatch* parent process must signal the child processes based on the actions required by the received call.

All "circuit switch" voice interfaces (labeled cs_pt_#") are directly sent into the "voice_dispatch" module. This module either forwards the voice data to the VoIP protocol (i.e., H.323/SIP) to be translated appropriately or sent back to another "circuit switch" interface. Secondly, dual 10/100Mbps Ethernet interfaces (labeled eth_pt_#_#") have been added to the standard NETWARS IP stack to give the Voice Gateway access to the IP network. Finally, dual ATM interfaces (labeled atm_pt_#_#") have been added to the standard NETWARS ATM/IP stack to give the Gateway access to ATM networks. Note that ATM interfaces are only available on the Media Gateway device and not on the N.E.T. SHOUTip device.

The remaining circuit interfaces (labeled "cir_intf_#") are out of scope for this document. These interfaces provide the necessary SAToP functionality for the N.E.T. SHOUTip device only. Please refer to the NETWARS CTP Model User Guide for further details. Essentially, this functionality works similar to the Juniper CTP model.
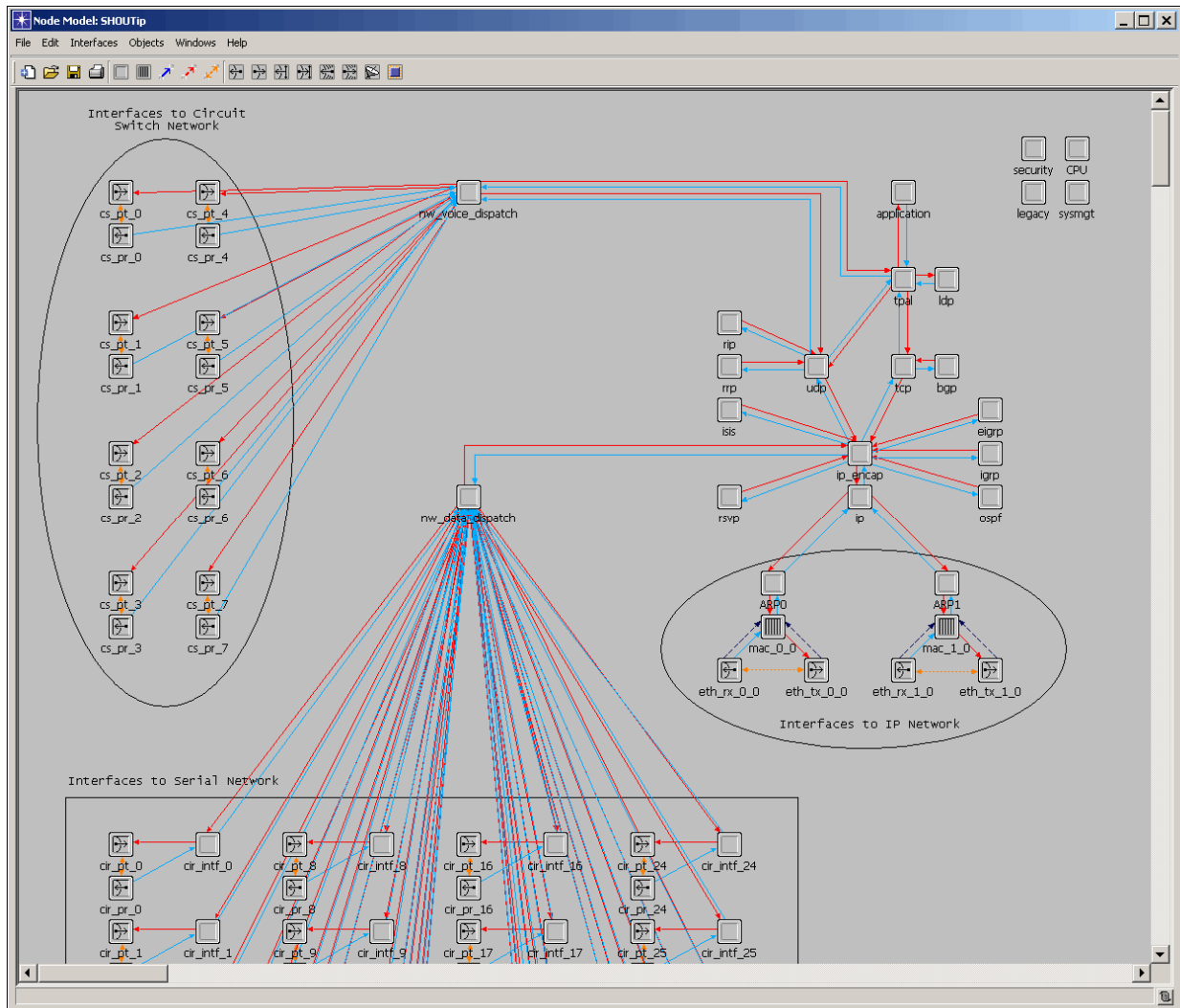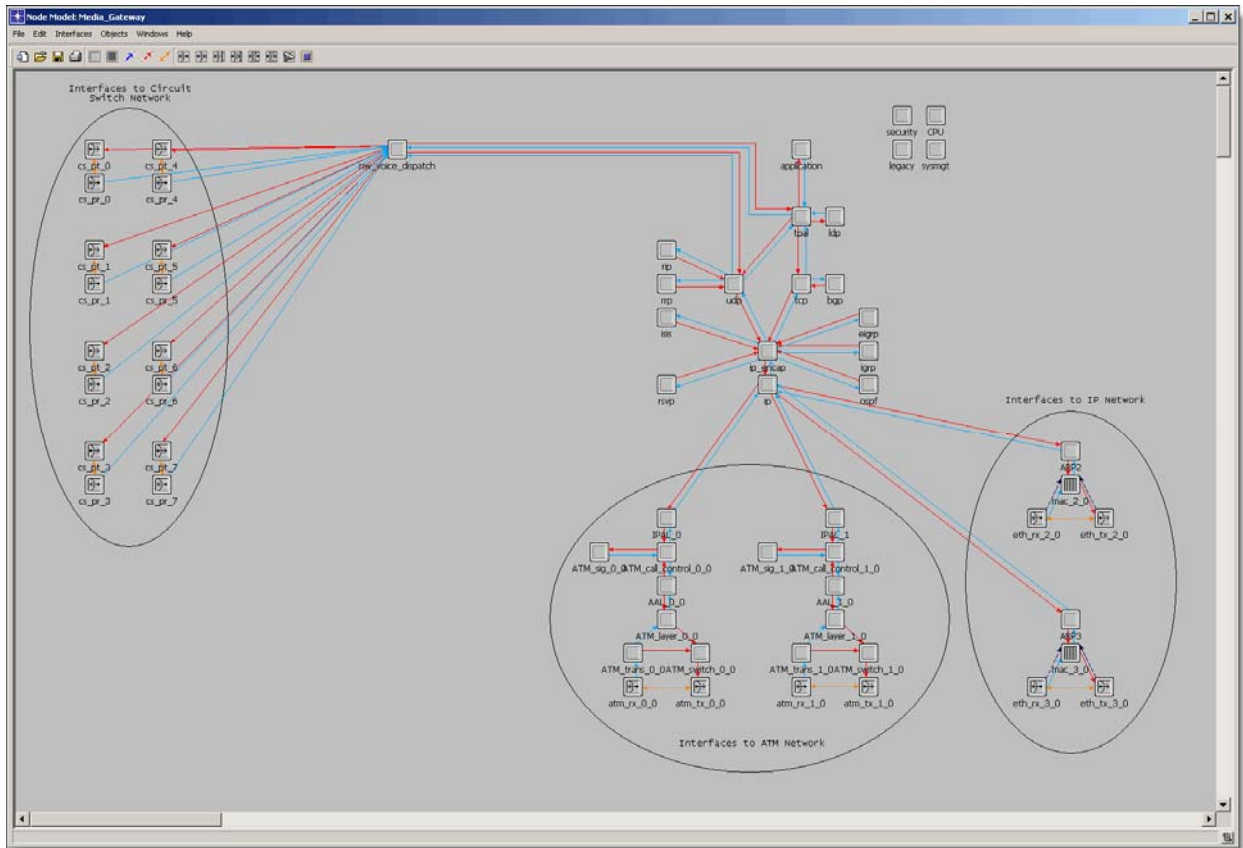
Figure 42: SHOUTip Node Model

Figure 43: Media Gateway Node Model

4.2   Device Creator

Currently, there is no device creator support with the Voice models in NETWARS.