



WinEyeQ

Revision 1.7.1

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WinEyeQ User's Guide

Table of Contents

Introduction.....	6
Version 1.5.0 Summary	7
New Definitions	7
Version 1.5.1 Summary	8
New Features.....	8
Version 1.5.2 Summary	10
New Features.....	10
Version 1.5.3 Summary	11
New Features.....	11
Version 1.5.4 Summary	12
New Features.....	12
Version 1.7.0 Summary	13
New Features.....	13
Version 1.7.1 Summary	14
Installation Types.....	15
WinEyeQ on CD-ROM	15
WinEyeQ via E-Mail	15
WinEyeQ via the Internet	15
WinEyeQ Installation.....	16
WinEyeQ Install Screen 1	16
WinEyeQ Install Screen 2 - Beginning the Installation	17
WinEyeQ Install Screen 3 - Beginning the Installation	18
WinEyeQ Install Screen 4 - End-User License Agreement	19
WinEyeQ Install Screen 5 - Readme Information	20
WinEyeQ Install Screen 6 - Customer Information	21
WinEyeQ Install Screen 7 - Destination Folder	22
WinEyeQ Install Screen 8 - Ready to Install.....	23
WinEyeQ Install Screen 9 - Installing WinEyeQ	24
WinEyeQ Install Screen 10 - Installation Complete.....	25
WinPcap Installation.....	26
WinPCap Install Screen 1 - Welcome to the installation Wizard.....	26
WinPcap Install Screen 2 - End-User License Agreement.....	27
WinPcap Install Screen 3 - Installation Progress	28
WinPcap Install Screen 4 - Installation Complete	29
Installation Notes.....	30
Running WinEyeQ for the First Time	31

WinEyeQ User's Guide

Obtaining the WinEyeQ Authorization Code	31
Transferring a License	32
Step One - Import License, Media Initialization	34
Step Two - Export License	36
Step Three - Install Exported License	40
License Transfer Instruction Chart	43
Selecting the Network Adapter	45
WinEyeQ User Interface	45
Data Scopes™	45
Network Monitor View	46
Network Bandwidth Consumption; top view	47
VoIP Bandwidth Consumption; level 2	47
Audio Bandwidth Consumption; level 3	48
G.711 Bandwidth Consumption Histogram; level 4	48
G.711 Bandwidth Consumption Histogram; zoomed.....	49
Navigational Tips.....	50
User Interface: Step-By-Step	50
The Network Monitor View	51
Network Summary	52
Network Details.....	52
VoIP Summary.....	55
Media Summary.....	55
Active Calls View.....	56
Call Summary	58
Call Flow.....	61
Call Trace	61
Call Metrics	62
Audio Summary	63
Audio Details.....	63
Audio QoS	65
Video Summary	67
Video Details.....	68
Data Details	70
RTCP Summary.....	72
RTCP XR Summary.....	74
DTMF Summary	75
Recent Calls View	76
Call Summary	78
Call Flow.....	81
Call Trace	81
Call Metrics.....	82
Audio Details.....	83
Audio QoS	85

WinEyeQ User's Guide

Video Details.....	87
Data Details	89
RTCP Summary.....	90
RTCP XR Summary.....	92
DTMF Summary	94
Recent Errors View	95
User Alerts View.....	97
User Alarms View.....	98
User Watches View.....	100
Endpoints View.....	102
Endpoint Summary and Recent Call History.....	103
Top Talker.....	104
Other Audio View	105
Other Audio Sub-Views	106
Other Video View	107
Other Video Sub-View	108
Registrations View.....	109
Registration Flow.....	110
Registration Trace	111
Registration Info.....	111
WinEyeQ Menu Commands	112
File Menu	112
Edit Menu.....	113
Capture Menu.....	117
Record Menu.....	117
View Menu	118
Help Menu.....	120
Toolbar Shortcuts.....	121
Selecting the Network Adapter and Packet Capture Filter	122
Configuration Settings.....	126
Logging	126
Calls	128
Protocol Filters	130
Reports.....	131
Preferences	134
Endpoints	136
Advanced	137
QoS.....	139
Display Filters	141
Alerts and Alarms.....	142
Categories.....	142
SNMP.....	144
Watches.....	146

WinEyeQ User's Guide

Capturing Calls	147
Overview	147
Capture - Calls with Errors	147
Capture - Watched Calls	147
Capture - Media Streams from Watched Calls	148
Capture - Rogue Streams	148
Recording Calls.....	149
Overview	149
Record - Watch Calls	149
Command Line Interface.....	150
WinEyeQ.cfg	151
Reports.cfg.....	152
AlertsAlarms.cfg	153
Watches.lst.....	154
Addresses.flt.....	155
Creating Custom Scenarios	156
Running Custom Scenarios.....	156
Creating and Running .bat files.....	157
WinPcap License	158
Appendix A	159
Theoretical Maximum MOS Scores and R Factors	159
Appendix B	160
Sample SNMP traps.....	160
Appendix C	161
Call Scoring - Letter Grades	161

Introduction

The WinEyeQ VoIP call monitor and protocol analyzer is the ideal tool for anyone who needs to monitor Voice and Video over IP calls and Voice quality, detect errors in VoIP traffic, debug signaling problems or capture media streams. WinEyeQ's intuitive user-interface makes setup and operation a snap.

With WinEyeQ you view your network traffic in an intuitive manner. From network overview to media stream and protocol details, each piece of information is presented in context. WinEyeQ's analysis does not stop at the call flow level; however, it provides unparalleled analysis of each individual call component making difficult diagnostics simple.

WinEyeQ is designed for the advanced 32 bit Windows operating systems. The following operating systems are supported:

Windows 2000 Professional, Windows 2000 Advance Server, Windows XP Professional, Windows 2003 Server.

WinEyeQ's capabilities automatically scale with the hardware on which it is installed.

Minimum recommended configuration:

- 2.4 MHz Pentium 4 Processor
- 512MB Ram
- 60 GB hard drive
- 1280x1024

WinEyeQ is optimized for 1280 x 1024 displays.

The WinEyeQ software is copy protected and is licensed for use on a single machine. Please make sure that you install WinEyeQ on the machine with which you intend to use it. Installation of WinEyeQ on multiple machines is not possible without authorization from Touchstone.

The following pages will demonstrate how to install, setup, and get started with WinEyeQ. The next session is an overview of the latest additions.

Version 1.5.0 Summary

New Definitions

Component

A tangible, measurable, or quantifiable element of an environment.

Data Scope

A graphical representation of a logical group of components organized in a “zoom-able” or variable-power fashion. Depths vary based upon logical grouping level, ranging from telescopic through microscopic.

Histogram

This is a historical representation of a component's value.

Anonymous or Rogue Media Streams

Any audio or video stream transmitted via RTP that WinEyeQ does not associate with an active or watched call.

Comments

Initially conceived as strictly a VoIP analyzer with monitoring capabilities; WinEyeQ is evolving into a comprehensive core component for monitoring, analyzing and diagnosing today's converging networks. The addition of Data Scope presentations and support for anonymous or “Rogue” media streams provide both breadth and depth to WinEyeQ's capabilities.

New Features

Continuing with WinEyeQ's “VoIP-centric” approach, Data Scopes provide a dramatic, immediately intuitive picture of the VoIP and non-VoIP components of your network. These groupings represent a broadening of scope for WinEyeQ, enabling engineers to have a clear, high-level picture of the network. This capability allows them to quickly identify and detect anomalies while simultaneously providing the ability to drill-down on any component at any time.

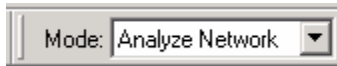
Another, much-requested capability makes its debut in 1.5.0. WinEyeQ now supports the detection, tracking, quantifying and qualifying of “Rogue” RTP media streams for both audio and video. These streams are defined as any RTP-based media connections that have been detected outside of analyzed capability exchanges. This new capability allows WinEyeQ to be used in any media-rich environment to analyze, quantify and qualify both audio and video streams regardless of the signaling protocol used to provide session control.

Version 1.5.1 Summary

New Features

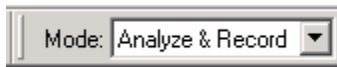
This version of WinEyeQ contains a Mode of Operation feature. There are three Modes of operation:

Analyze Network



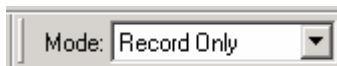
This is the normal mode of operation for WinEyeQ. Packets are captured either from a live network or read in from files of previously captured data. VoIP calls are assembled and displayed on the screen. Individual call metrics are displayed. Various reports may be generated and network statistics are calculated and displayed. This is the suggested mode of operation.

Analyze and Record



This mode of operation is the same as the Analyze Network mode except that a trace file of all the packets received by WinEyeQ is also created. The user can specify the name of the file and its format, either native WinEyeQ or WinPcap, and the maximum file size. This mode of operation could be used create a trace of an error condition that includes the events that lead up to that condition.

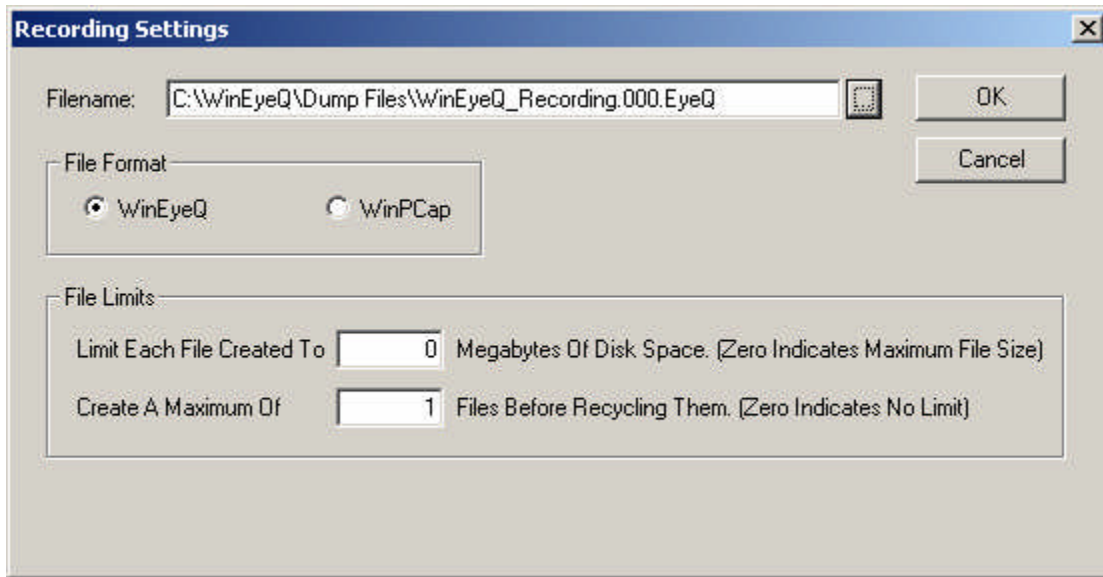
Record Only



This mode of operation only creates a trace file of the packets received by WinEyeQ. The packets are not examined and nothing is displayed on the screen. The user can specify the name of the file and its format, either native WinEyeQ or WinPcap, and the maximum file size. This mode of operation can be used to create trace files for offline analysis.

WinEyeQ User's Guide

If you start WinEyeQ in Analyze & Record Mode or Record Only Mode, the following dialog will be displayed:



The image shows a Windows-style dialog box titled "Recording Settings". It has a standard title bar with a close button (X). The dialog contains three main sections: 1. "Filename:" with a text input field containing "C:\WinEyeQ\Dump Files\WinEyeQ_Recording.000.EyeQ" and a file icon button to its right. 2. "File Format" with two radio buttons: "WinEyeQ" (which is selected) and "WinPCap". 3. "File Limits" with two rows of controls. The first row is "Limit Each File Created To" followed by a text box containing "0" and the text "Megabytes Of Disk Space. (Zero Indicates Maximum File Size)". The second row is "Create A Maximum Of" followed by a text box containing "1" and the text "Files Before Recycling Them. (Zero Indicates No Limit)". At the bottom right of the dialog are "OK" and "Cancel" buttons.

Please enter the name of the file you want to create, the format for that file, the maximum size of each file that will be created, and the maximum number of files to create.

When WinEyeQ begins recording (using the above example), it will create up to 10 files in the WinEyeQ format, each up to 100 megabytes in size. The first file created will be named WinEye_Recording.000.EyeQ. If it reaches 100 megabytes in size, it will be closed and a new file named WinEye_Recording.001.EyeQ will be created. If that file reaches 100 megabytes, it will be closed and another file named WinEye_Recording.002.EyeQ will be created and so on up to a maximum of 10 files. When 10 files have been created, the file-naming scheme will start over again with WinEye_Recording.000.EyeQ.

Note: The trace files that WinEyeQ creates can grow very large especially if there is a lot of traffic on the network. Be sure to filter out as much unnecessary traffic as possible (See Selecting the Network Adapter and Packet Capture Filter).

Version 1.5.2 Summary

New Features

This version of WinEyeQ implements a new folder structure that separates reports, captures, traces, etc. out of the main WinEyeQ folder. At the present time the folder names are fixed and are as follows:

Audio Capture Files

Audio streams (Rogue or from Watched calls) that are captured will be created here. These streams are in their raw form, i.e. as if they were taken directly from the output of the CODEC that generated them.

Video Capture Files

Video streams (Rogue or from Watched calls) that are captured will be created here. These streams are in their raw form, i.e. as if they were taken directly from the output of the CODEC that generated them.

Trace Files

When the user selects 'Call Trace,' 'Watch Trace,' or 'Error Trace,' the trace files will be created here. These trace files are text files that show each packet of a call.

Reports

All user-selected reports will be created here.

Record Files

When the user selects "Record Watched Calls," the recorded files will be created here.

Dump Files

When the user selects "Analyze and Record" or "Record Only" in the "Mode" dropdown list, the recorded files will be created here.

Capture Files

When the user selects "Capture Watched Calls" or "Capture Calls With Errors," the recorded files will be created here.

Version 1.5.3 Summary

New Features

This version of WinEyeQ allows the user to capture rogue media streams to the disk by right clicking the stream on the Audio or Video channels screen, then select Start Capture or Stop Capture.

There is a new Alert / Alarm that detects duplicate media stream destinations, i.e. two different source endpoints sending to the same destination endpoint.

WinEyeQ will now not report a SIP response of 486 Busy as an error.

Version 1.5.4 Summary

New Features

WinEyeQ added support for H.264 Video and AAC LD Audio for H.323 calls.

There is a new tab on the Active Calls and Watched Calls screen that provides metrics for data (neither audio nor video) channels.

The select adapter screen now accepts a range of IP addresses.

A WinEyeQ user can now configure the jitter buffer and international QoS settings.

Support has been added to decode and analyze RTCP XR messages. A new RTCP XR report saves this information to the disk.

All WinEyeQ reports now support size, interval, and time of day constraints. The current report file will be closed and a new one opened when the report reaches a certain size (in megabytes), or periodically (hours and minutes) or at a certain time of the day (midnight for example).

There is a new idle timeout parameter on the Settings menu: Calls Tab. This parameter specifies the amount of time a call will be considered active when it is not transmitting data in one direction or the other. The other idle timeout parameter is for calls that are not transmitting in both directions.

Version 1.7.0 Summary

New Features

In past versions of WinEyeQ, there was 'Maximum Calls' configuration parameter. This parameter was the maximum number of calls and registrations that WinEyeQ would track and was based on the version of the application (Professional, Lite, and Demo). There are now two parameter values, first, 'Maximum Calls' which limits the number of calls and second, 'Maximum Registrations' which limits the number of registrations. These two parameters are independent of each other but are still limited by the version of the program. The maximum number of registrations is twice the number of call allowed.

There is now a 'Top Talkers' sub-tab on Endpoints screen. This tab shows the endpoints that have placed / received the most calls, have been connected the longest, and have used the most bandwidth.

WinEyeQ now has the ability to capture media streams in raw or packetized format. Previously the streams were only captured in the raw format. The packetized format preserves the RTP header information of each packet that is saved to the disk. Both of these file formats can be used by Touchstone's call generators to generate RTP test patterns or another application that can post process the media file.

A 'Recent Calls' tab has been added to the main screen. This screen shows the calls that have completed successfully after they have been removed from the Active Calls screen. Calls that are unsuccessful are still added to the Recent Errors screen.

This version of WinEyeQ now can generate SNMP traps. The traps are triggered by exceeding Alert / Alarm thresholds as configured by the user. See Appendix B for SNMP samples.

A new call scoring method has been added to WinEyeQ. This provides a simple way for the user to evaluate the quality of the call. Each call is given a letter grade and a numerical score. Each media stream present as well as the signaling time is factored together to for an overall call score. Please see Appendix C for scoring details.

An option has been added to the Reports tab that ensures that all report files will have unique names.

Version 1.7.1 Summary

Changes for Release 1.7.1

- Corrected problem using SNMP default IP address 255.255.255.255
- Replaced file extension *.cap with *.pcap in all file dialogs

Installation Types

WinEyeQ on CD-ROM

If you received WinEyeQ on CD-ROM, please use the following procedure:

- Insert the WinEyeQ CD in your CD-ROM drive.
- The installation program should start automatically. If it does not, use Windows Explorer to browse the CD and double-click the Setup.exe file.
- Continue to the next section.

WinEyeQ via E-Mail

If you received WinEyeQ via E-Mail, please use the following procedure:

- Double-click on the e-mail attachment.
- Select "Save to Disk" option and select a temporary folder to store the self-extracting file.
- Use Windows Explorer to browse to the folder in which you saved the self-extracting file.
- Double-click the self-extracting file. Select a folder with which to extract the files.
- Use Windows Explorer to browse to the folder you extracted the files in and double-click the Setup.exe file.
- Continue to the next section in this document.

WinEyeQ via the Internet

If you downloaded WinEyeQ via the internet, please use the following procedure:

- WinEyeQ's setup.exe is compressed using WinZip. Download wineyeq.zip and extract the setup.exe to a temporary location on the destination computer.
- Double-click on the Setup.exe file.
- Continue to the next section in this document.

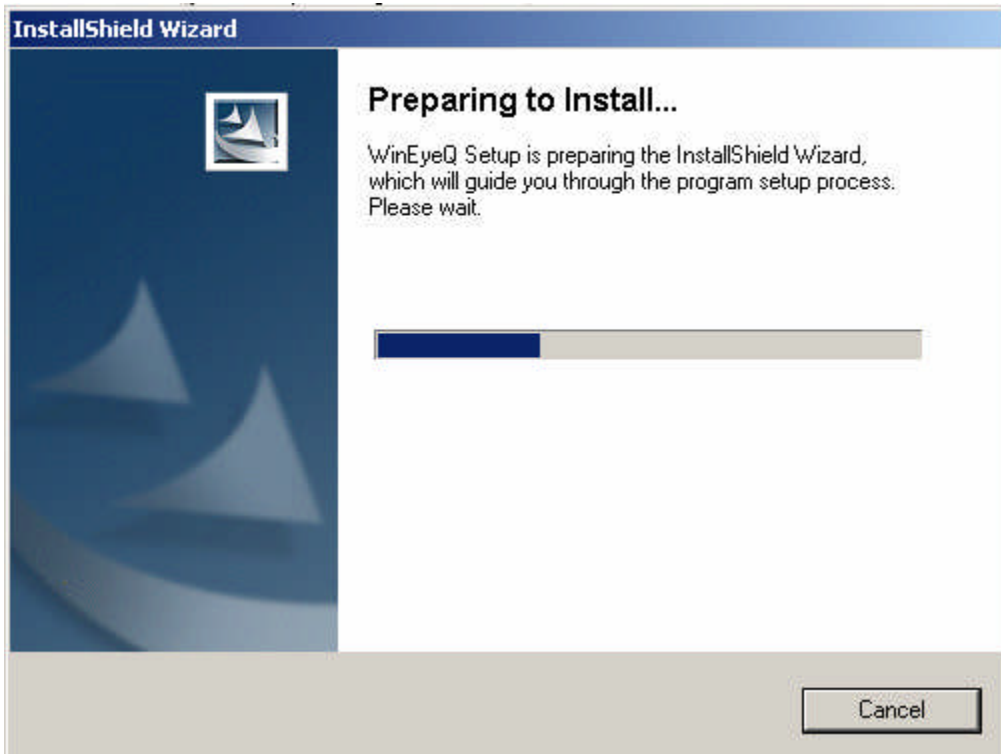
WinEyeQ Installation

The next few screens will appear during the installation process. Please follow the directions carefully using the “Next” button to navigate forward and the “Back” button to return to a previous page.

WinEyeQ Install Screen 1

Preparing Setup Wizard

Wait for the wizard to complete or press the “Cancel” to quit the installation.



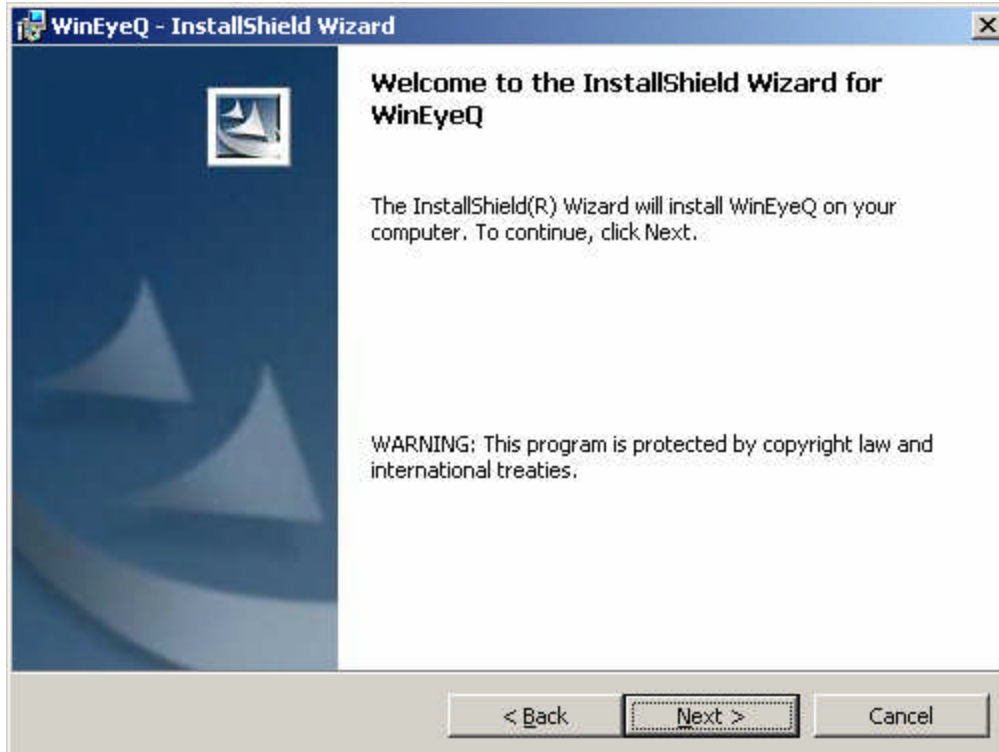
WinEyeQ Install Screen 2 - Beginning the Installation

Press the "Next" button to continue the installation or "Cancel" to quit.



WinEyeQ Install Screen 3 - Beginning the Installation

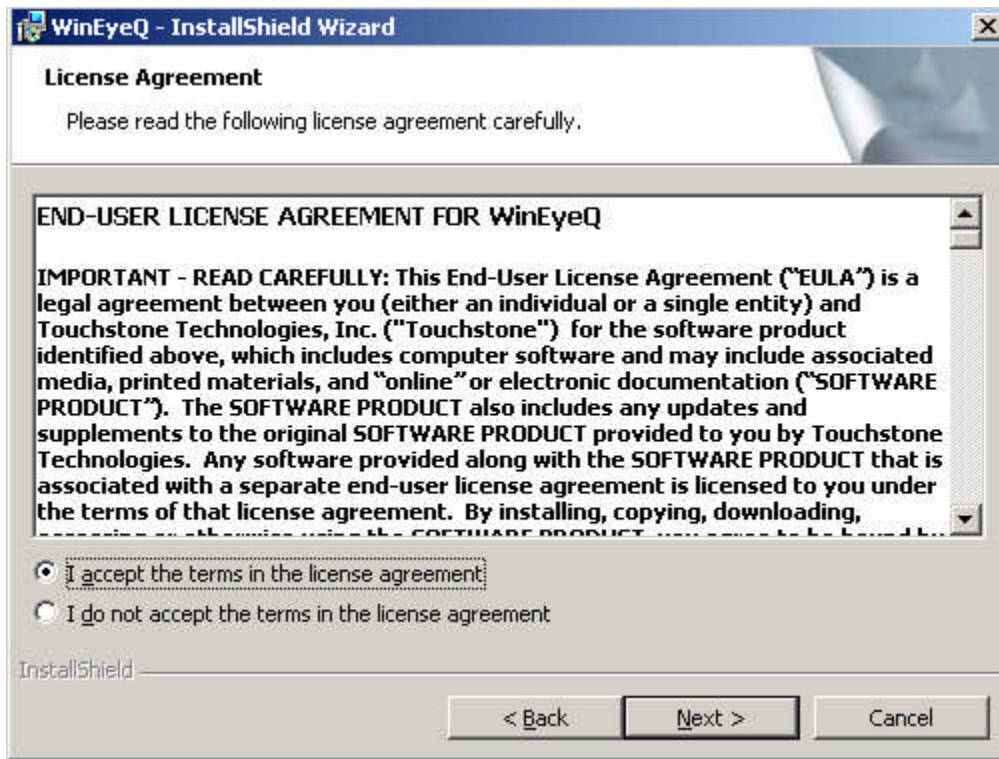
Press the “Next” button to continue the installation or “Cancel” to quit.



WinEyeQ Install Screen 4 - End-User License Agreement

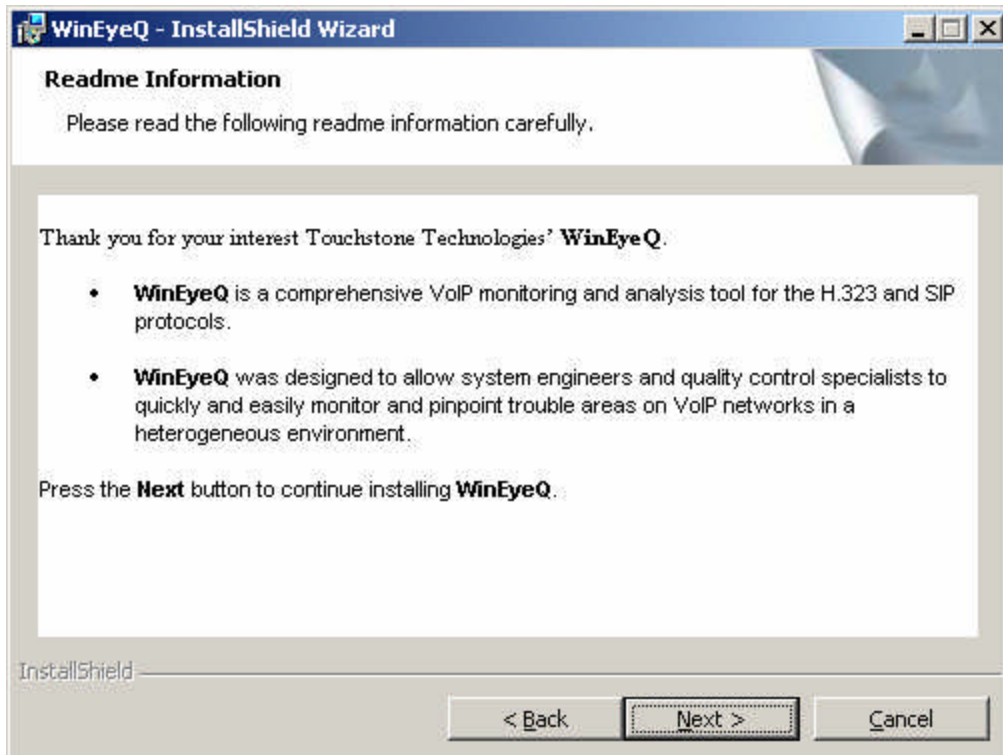
Carefully read the End-User License Agreement. If you accept the terms, select the "I Accept" option, if you do not; select the "I do not accept" option.

Press the "Next" button to continue the installation or "Cancel" to quit.



WinEyeQ Install Screen 5 - Readme Information

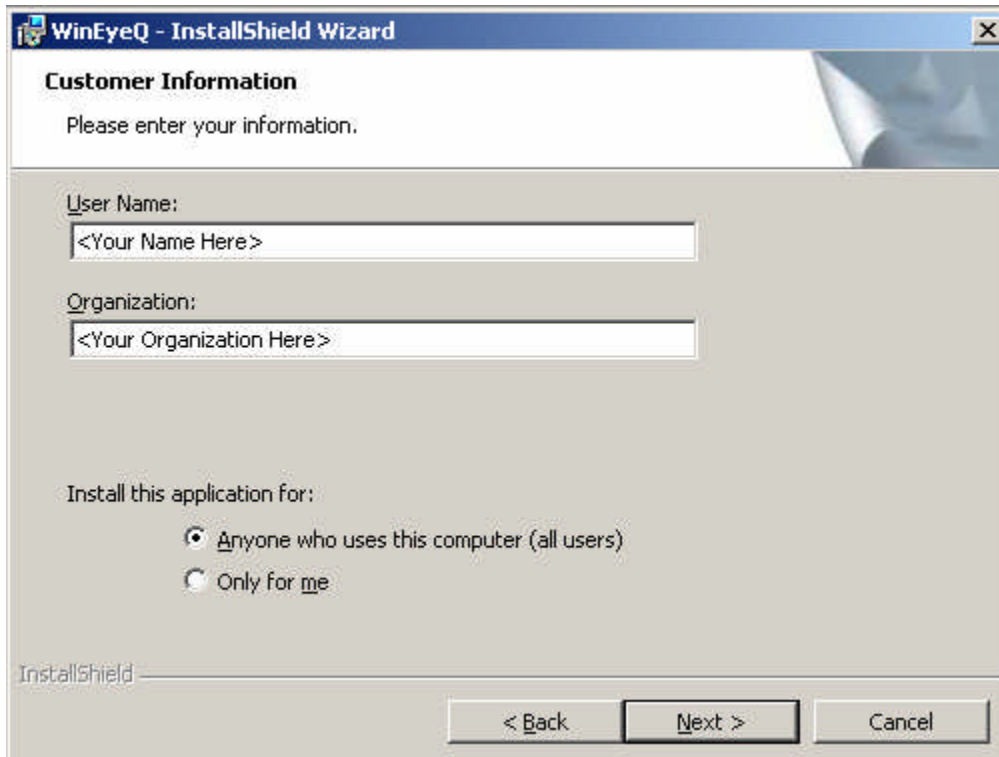
Press the “Next” button to continue the installation or “Cancel” to quit.



WinEyeQ Install Screen 6 - Customer Information

Please fill in your customer information and select the appropriate security option.

Press the “Next” button to continue the installation or “Cancel” to quit.

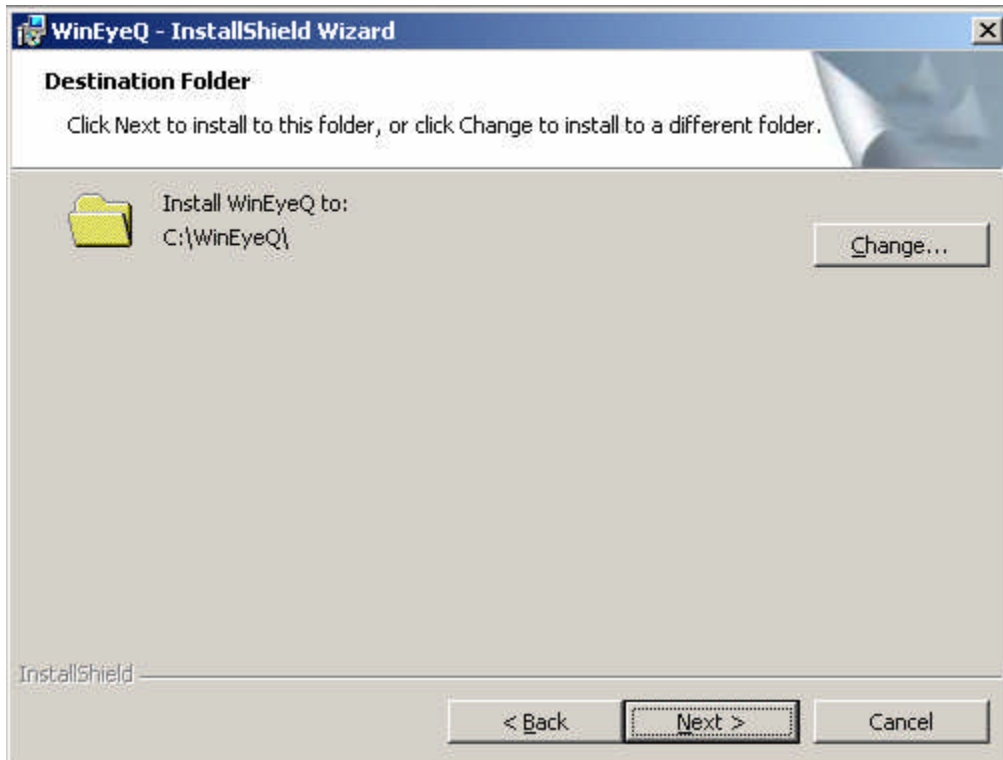


The image shows a Windows-style installation window titled "WinEyeQ - InstallShield Wizard". The window has a blue title bar with standard minimize, maximize, and close buttons. The main content area is titled "Customer Information" and contains the instruction "Please enter your information." Below this, there are two text input fields: "User Name:" with a placeholder "<Your Name Here>" and "Organization:" with a placeholder "<Your Organization Here>". Further down, under the heading "Install this application for:", there are two radio button options: "Anyone who uses this computer (all users)" which is selected, and "Only for me". At the bottom left, the "InstallShield" logo is visible. At the bottom right, there are three buttons: "< Back", "Next >", and "Cancel".

WinEyeQ Install Screen 7 - Destination Folder

Please select the folder in which you would like to install WinEyeQ and its components.

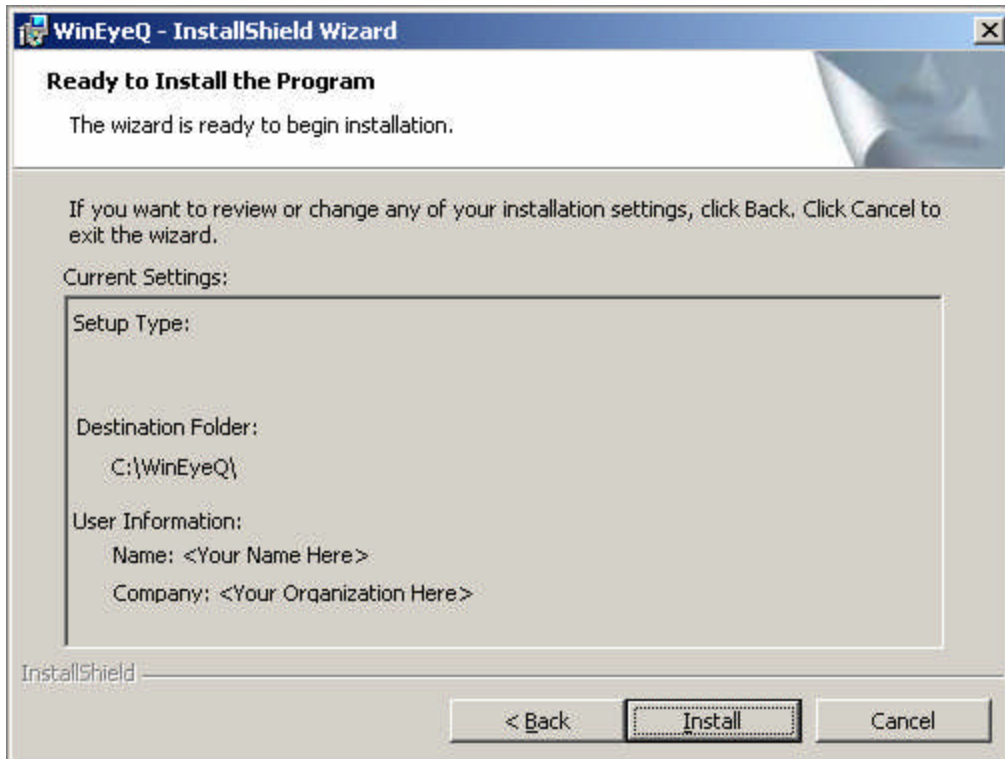
Press the “Next” button to continue the installation or “Cancel” to quit.



WinEyeQ Install Screen 8 - Ready to Install

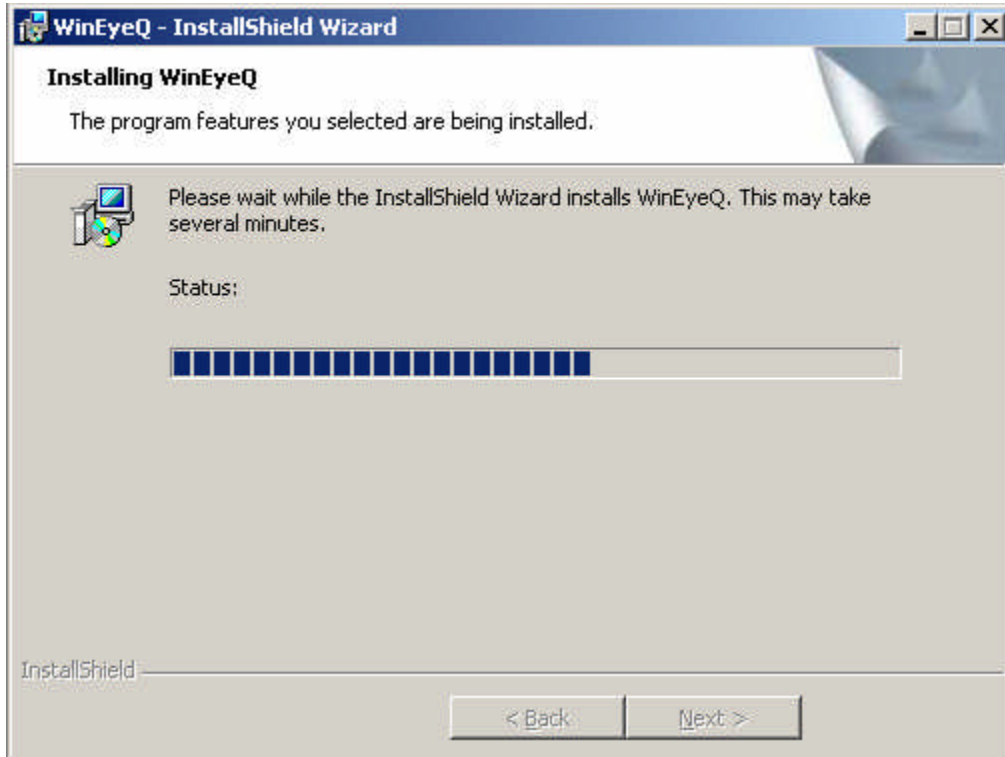
Please review the information, if you need to correct anything, use the “Back” button to navigate to the appropriate screen, make your changes and use the “Next” button to advance back to this point.

Press the “Install” button to continue the installation or “Cancel” to quit.



WinEyeQ Install Screen 9 - Installing WinEyeQ

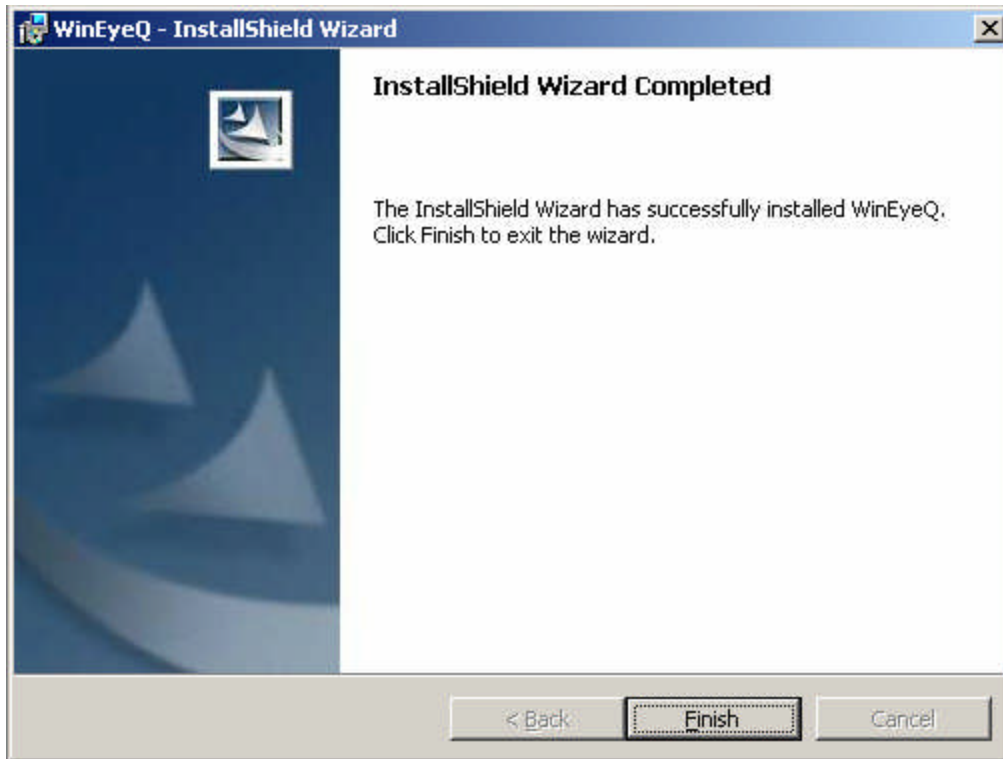
This screen will appear during the installation to inform you of the progress. Typically this screen will only appear for a very brief period of time.



WinEyeQ Install Screen 10 - Installation Complete

This screen will appear at the completion of the installation process. Any errors that may have occurred will be reported at this time. Should you encounter any errors, please contact Touchstone for technical assistance at +215.672.6550 or support@touchstone-inc.com.

Press the “Finish” button to complete the installation.

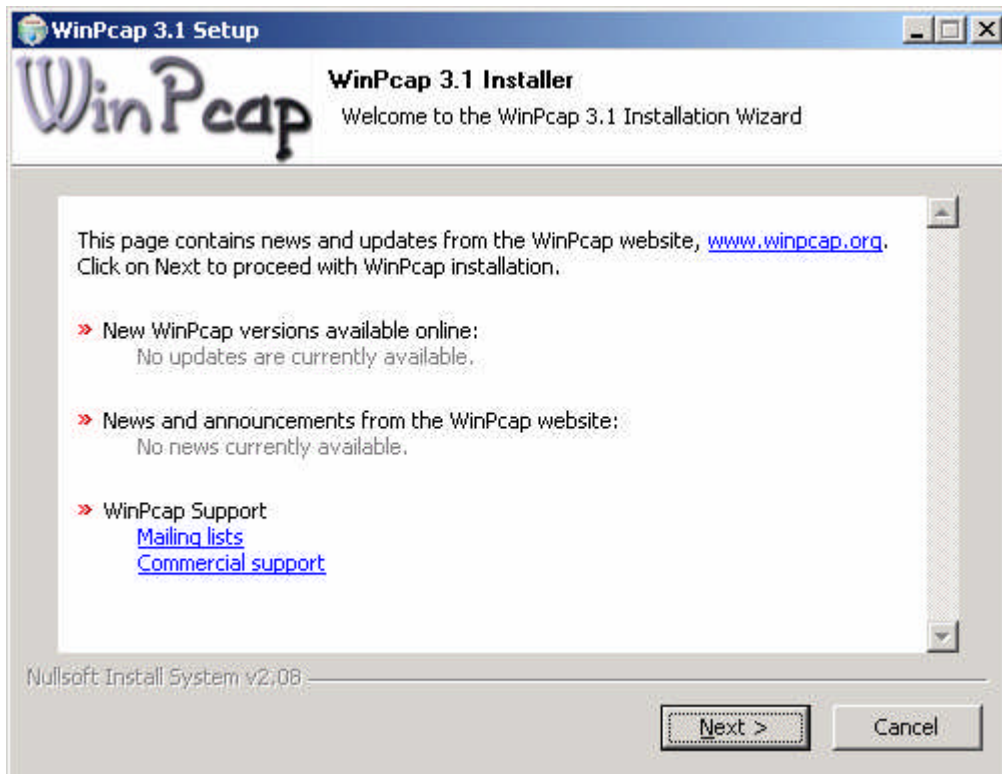


WinPcap Installation

Before the installation is complete, it is necessary to install the WinPcap driver. If you have installed other products that use this driver (such as Ethereal), you will probably need to restart the computer after installation. The following screens will appear during the WinPcap installation process. Please follow the directions carefully using the “Next” button to navigate forward and the “Back” button to return to a previous page.

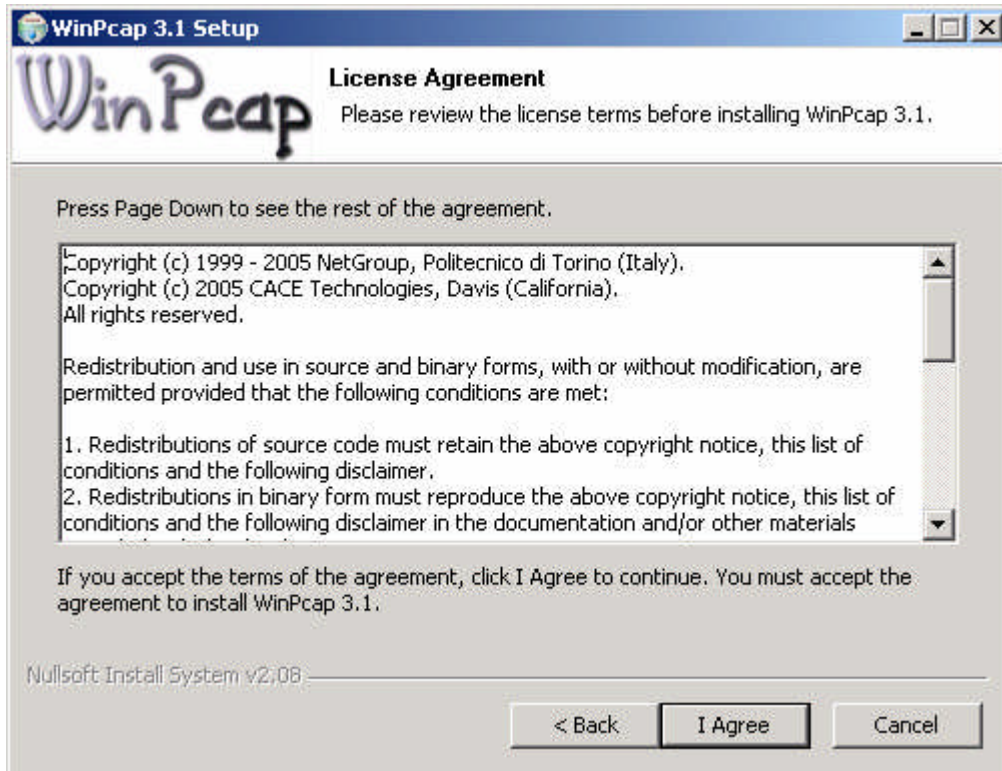
WinPCap Install Screen 1 - Welcome to the installation Wizard

Press the “Next” button to continue or the “Cancel” to quit the installation.



WinPcap Install Screen 2 - End-User License Agreement

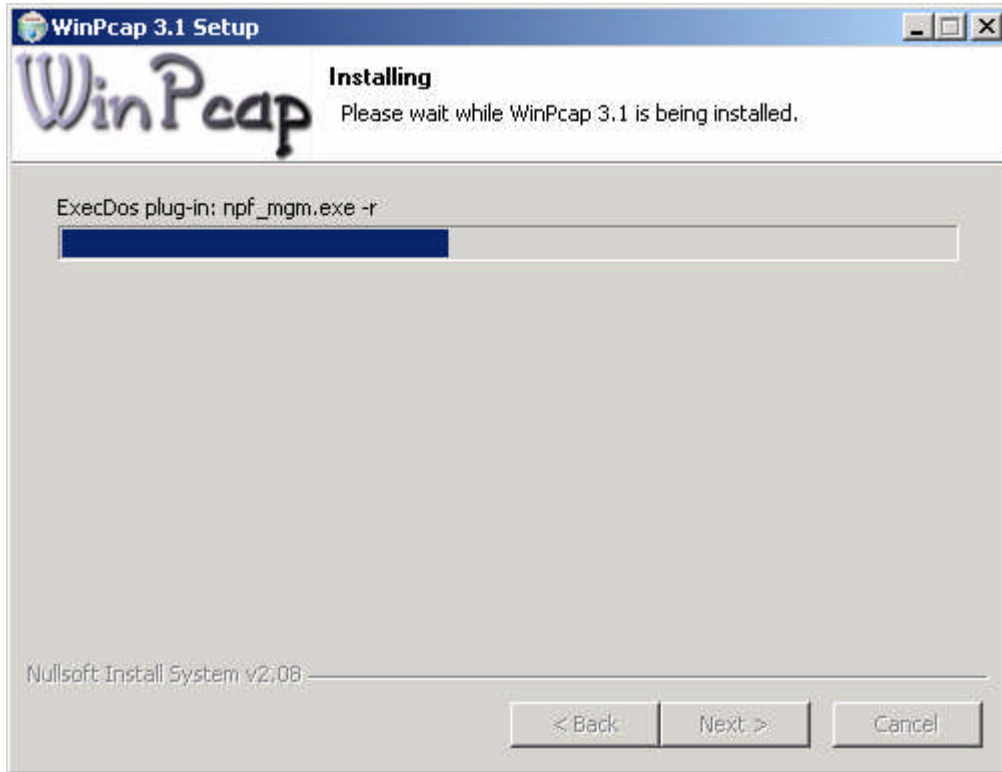
Carefully read the License Agreement. If you accept the terms, press the “I Agree” button, if you do not, press the “Cancel” button.



Press the “Next” button to continue the installation or “Cancel” to quit.

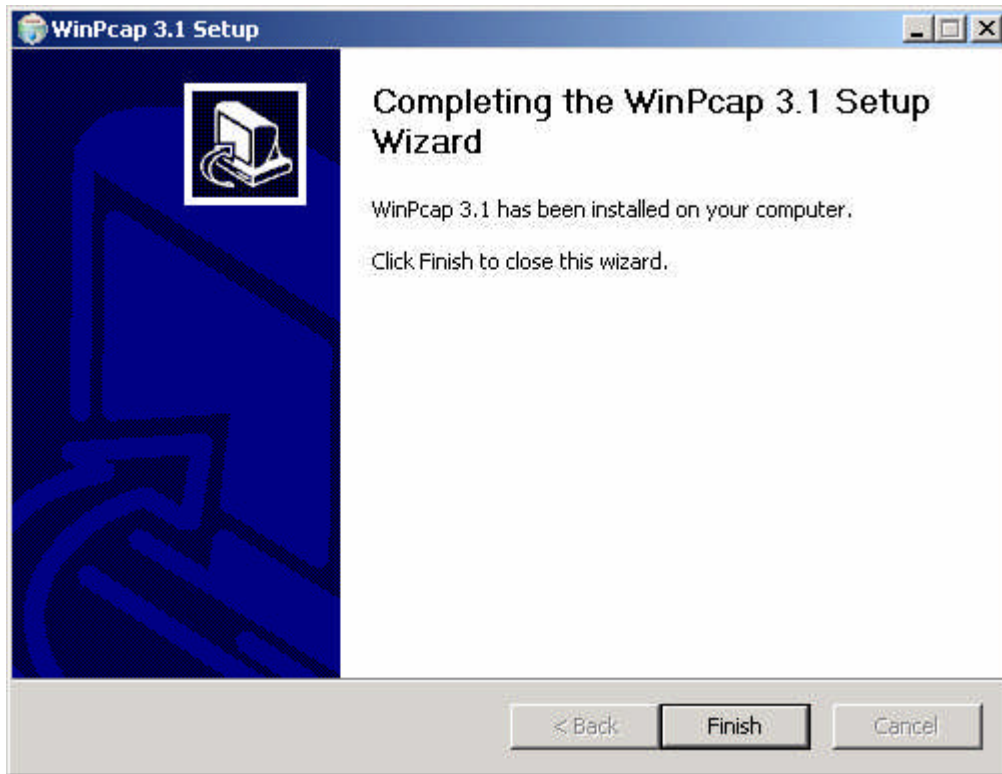
WinPcap Install Screen 3 - Installation Progress

This screen will appear during the installation process.



WinPcap Install Screen 4 - Installation Complete

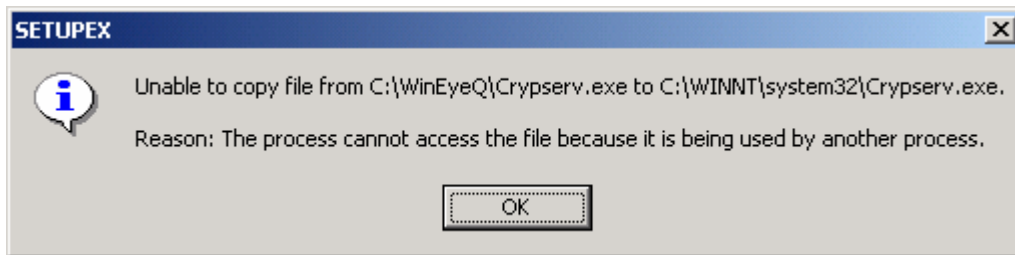
The following screen will appear at the completion of the WinPcap installation.



Installation Notes

The installation process will create a shortcut on your Windows desktop for the WinEyeQ application. The “Start” menu’s “Programs” section will also contain an entry for WinEyeQ. You may use either of these to run your WinEyeQ application.

If there are other applications from Touchstone Technologies installed on your PC, a message similar to the one below may appear at the conclusion of the installation.



You can safely ignore this message.

Running WinEyeQ for the First Time

WinEyeQ software is copy protected and is licensed for use on a single machine. The first time you run WinEyeQ, you will be provided with a site code. You must contact Touchstone in order to obtain the authorization code to enable the software.

Once the software is authorized, it may not be installed on any other machines without a new authorization code from Touchstone. If you have installed the software on a machine in error, do not authorize that installation. Re-install it on the appropriate machine prior to contacting Touchstone for the authorization code.

Obtaining the WinEyeQ Authorization Code

When you first run WinEyeQ the following authorization dialog will appear:



In the field labeled "Site Code" a series of numbers and letters will appear. To authorize the application, contact Touchstone with the **exact** value of the site code field. Touchstone will provide the code to enter in the "Authorization Code" field. You must enter this **exactly** as it is provided to you in order to enable the software. It is strongly suggested that you 'copy' the site code into an email that you send to Touchstone, and then 'paste' the authorization code from the email you receive from Touchstone. Once you have enabled the software, you are just moments away from being able to construct your first test scenarios!

Transferring a License

The method of transferring a license is the same for all Touchstone Technologies products. For demonstration purposes WinEyeQ will be used to explain the license transfer procedure.

At the time of installation there are two options for licensing WinEyeQ. The first is to have a new key issued from Touchstone Technologies, and the second is to transfer a license from an existing WinEyeQ application to the newly installed version of WinEyeQ. Touchstone's software licenses are fully transferable from PC to PC within a customer's physical location. To transfer a license to a different location, please contact Touchstone Technologies at (215) 672-6550.

A floppy diskette or USB memory device is required to transfer a license.

There are three basic steps in transferring a license:

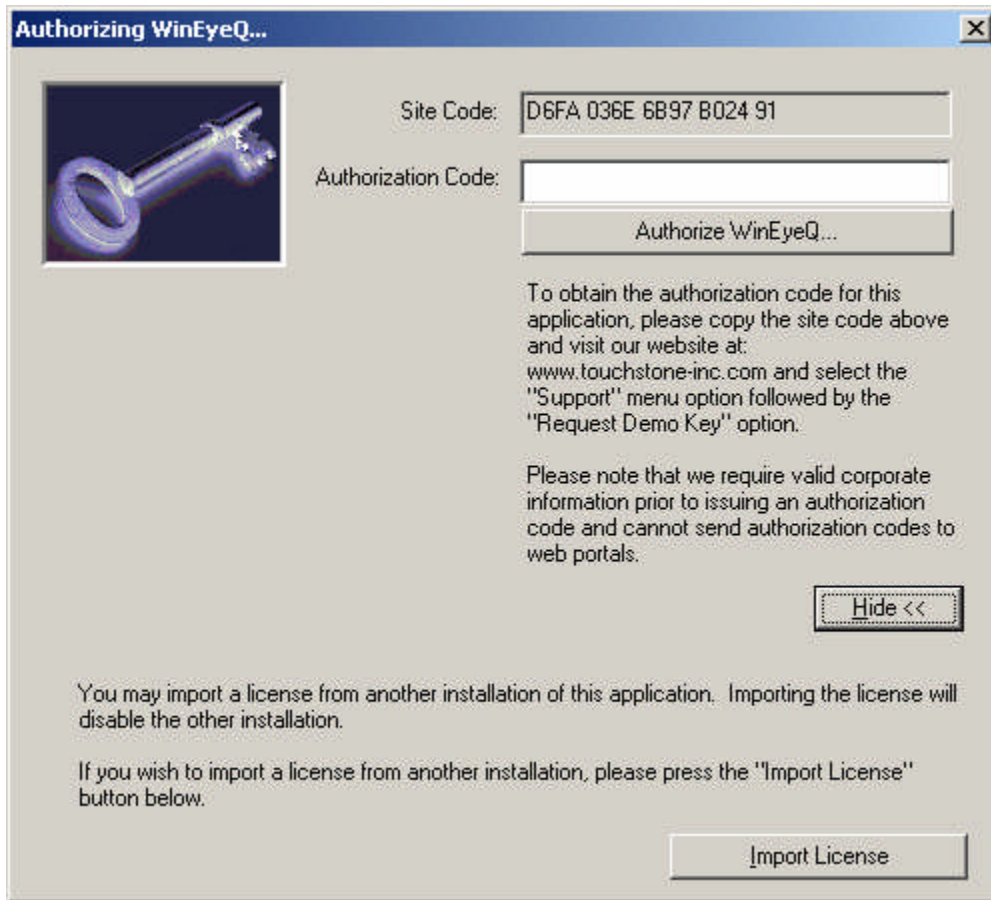
- Initialize transfer media on the PC with newly installed WinEyeQ.
- Export license from the PC with the originally installed WinEyeQ.
- Import license to the PC with newly installed WinEyeQ.

Note: Touchstone Technologies licenses will have to be re-issued if:

- The original installation directory of WinEyeQ is:
 - Copied or moved to a new directory on the original PC.
 - Copied or moved to a different PC.
 - Renamed
- One of the hidden files (deltaps.ckn or deltaps.inf) is deleted or modified.
- The license service (crypserv.exe) is stopped or uninstalled.

WinEyeQ User's Guide

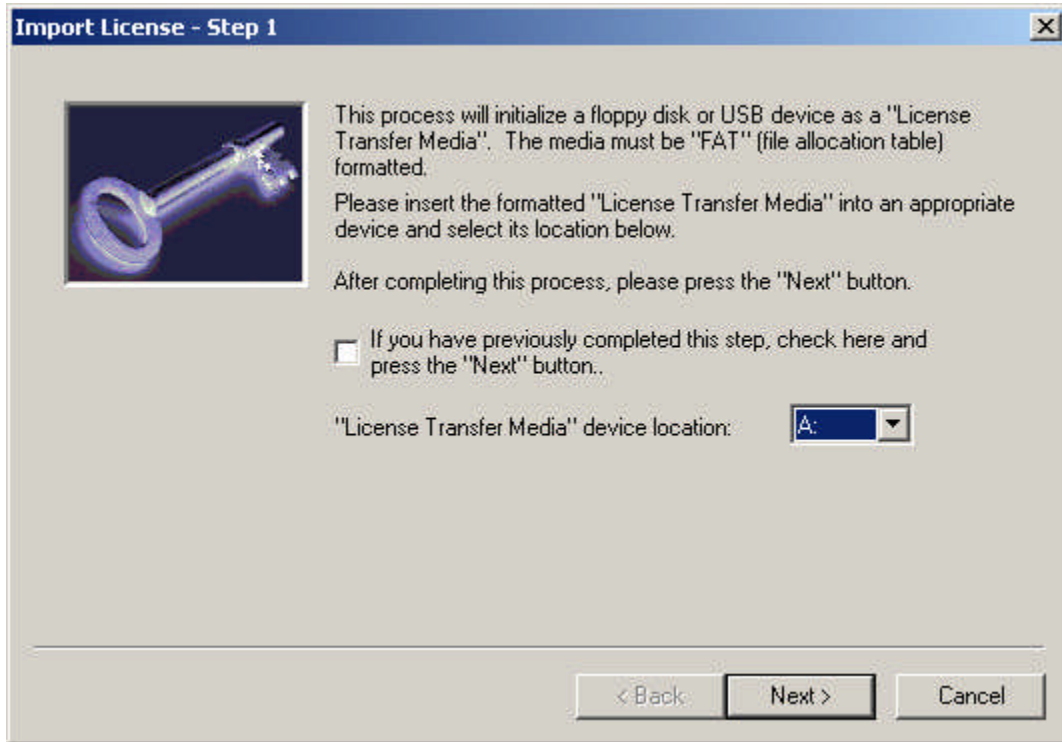
After a new installation is finished and the application is run for the first time, an 'Authorizing WinEyeQ' screen will appear. By clicking on the 'Advanced' button, an expanded dialog will be displayed:



Press the 'Import License' button to begin the license transfer procedure.

Step One - Import License, Media Initialization

The first step of the 'Import License' transfer requires initialization of a diskette or USB device that will be used as the 'License Transfer Media'.

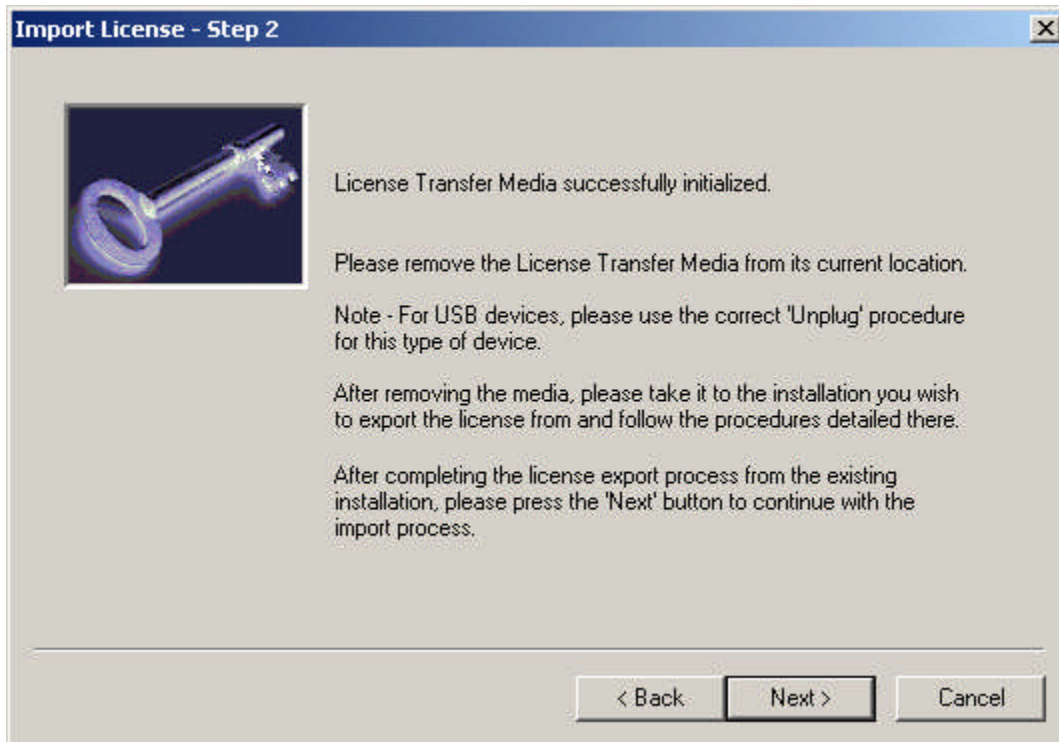


To initialize the transfer media, select the drive to be used as the transfer device, insert the transfer media and press the 'Next' button.

Note: If you have completed this step from a previous execution of WinEyeQ and already have the initialized transfer media, click the checkbox and then press the 'Next' button.

When step one is complete, the transfer media is initialized.

The 'Import License - Step 2' dialog will then appear:



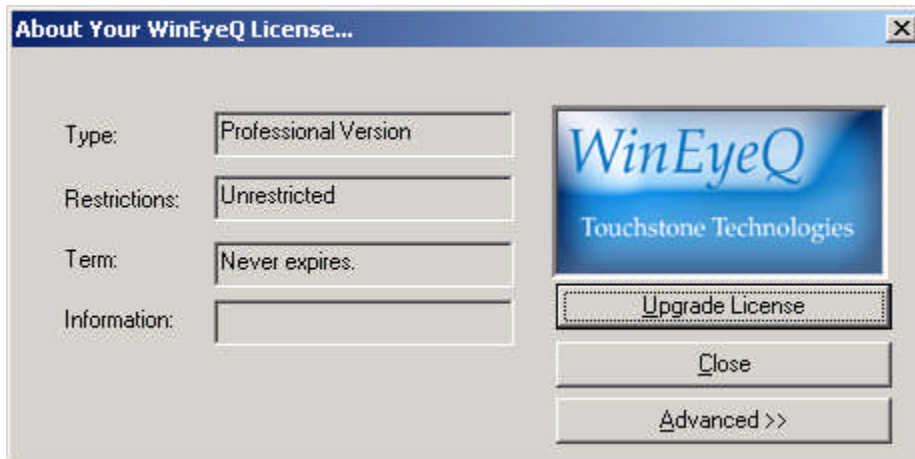
The next step is to eject or unplug the transfer media and take it to the PC that has the license you want to remove.

Note: For USB devices please follow the correct unplug procedure for your device.

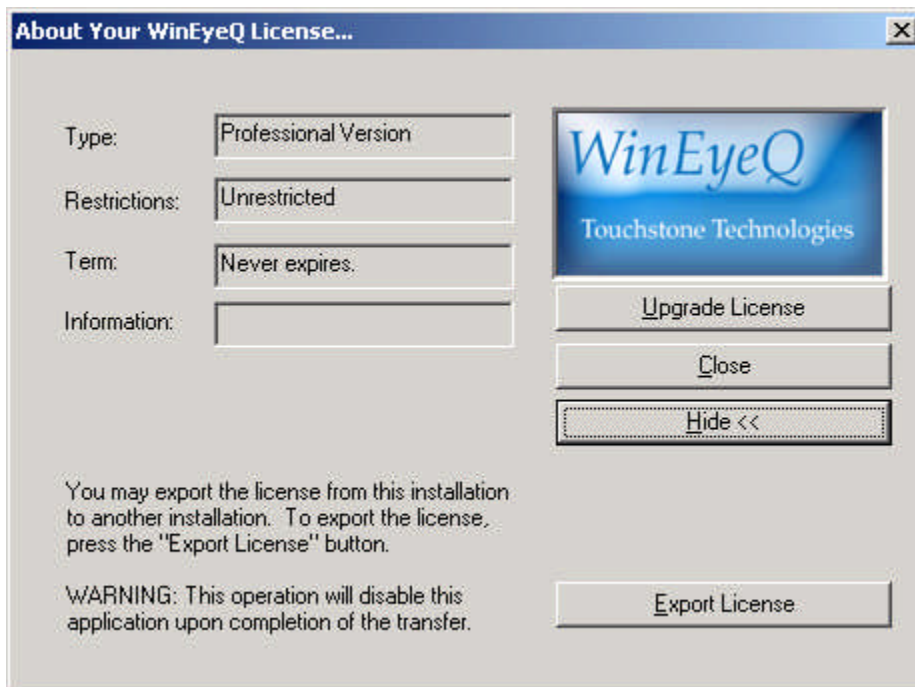
Step Two - Export License

On the PC that you have selected to remove the WinEyeQ license, click on the 'Help' menu and then select 'Licensing Information'.

The Following dialog will appear:



Next click on the 'Advanced' button to expand the dialog:



Now click on the 'Export License' button.

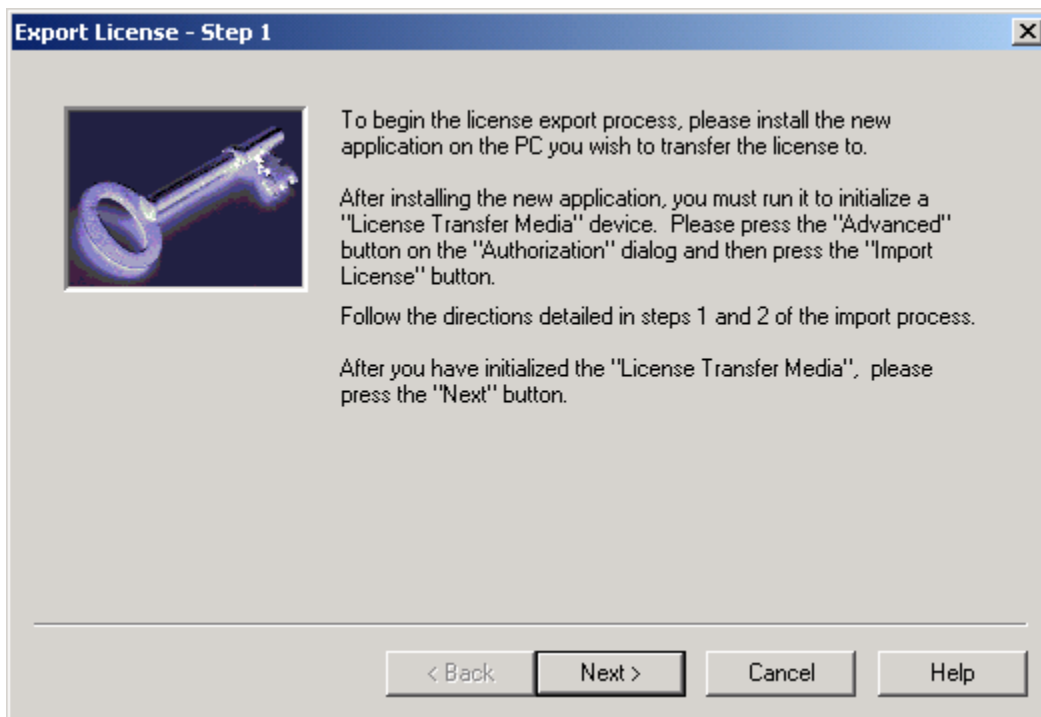
WinEyeQ User's Guide

A warning dialog will be displayed next. It instructs you to read the procedure carefully and that the version of WinEyeQ currently running will be disabled after the procedure is completed.

If you are certain you want to transfer this license, press 'Yes,' if not, press 'No'.

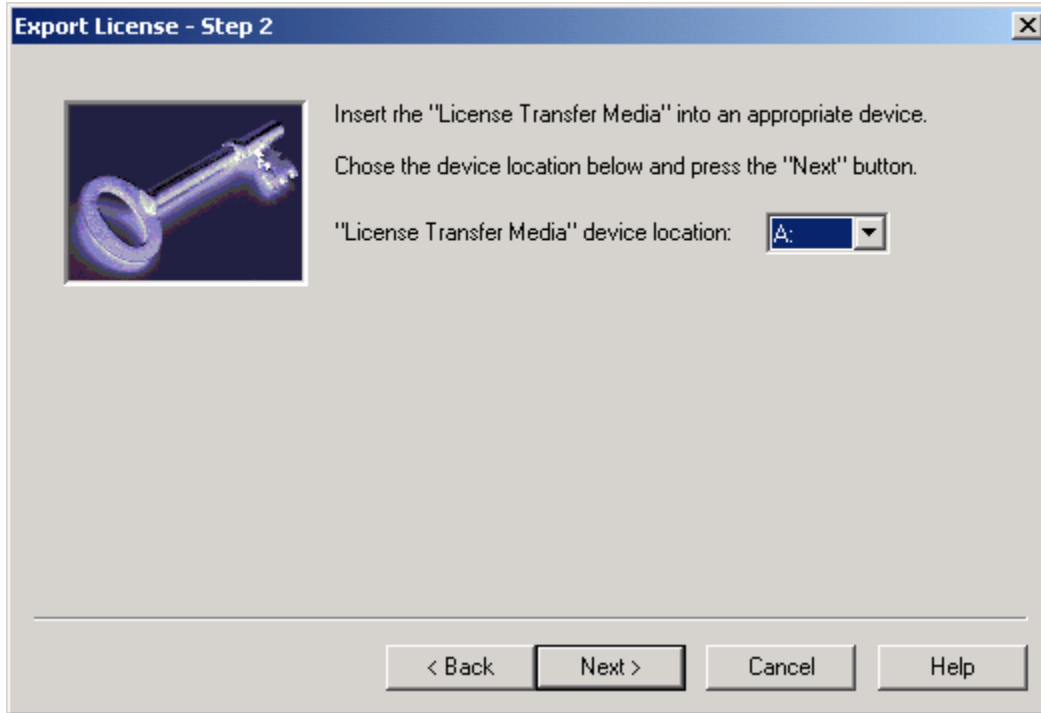
The existing WinEyeQ application will not be uninstalled nor will any WinEyeQ files be removed from the WinEyeQ directory, the software will simply be disabled. Later if you wish, you can re-enable the application with a new license from Touchstone or with a WinEyeQ license transferred from another PC.

Step one of the export procedure displays the following dialog:



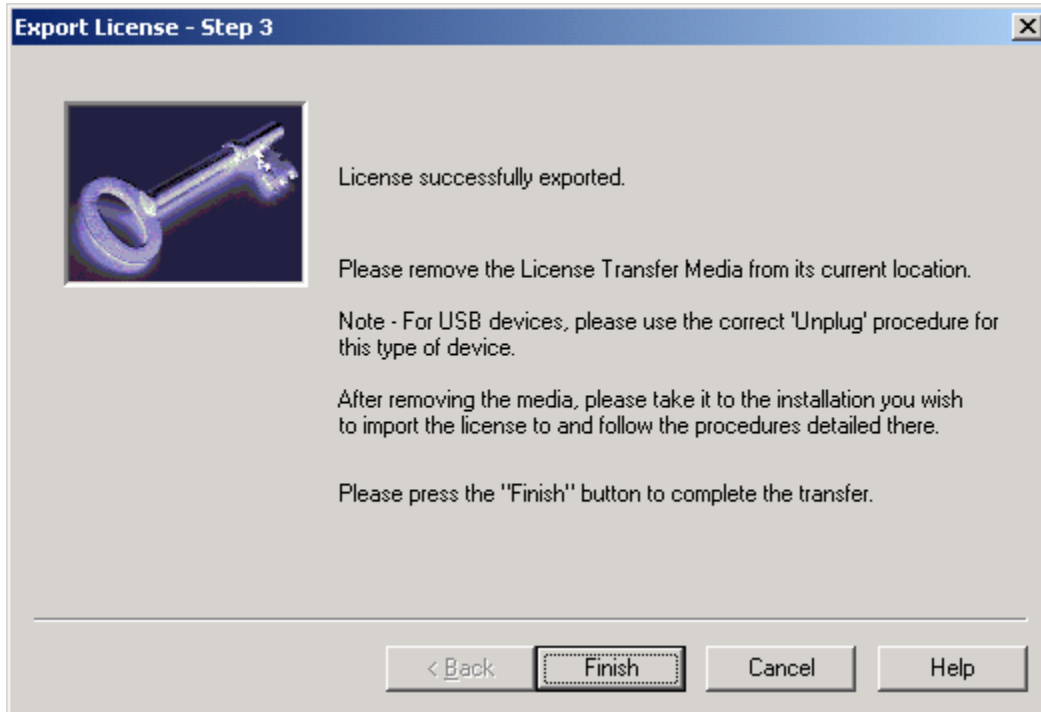
Click the 'Next' button.

Step two of the license export procedure displays the following dialog:



Insert the transfer media that was initialized from 'Step One – Media Initialization,' select the drive to be used as the transfer device and press the 'Next' button.

When the license has been successfully exported, the following dialog will appear:



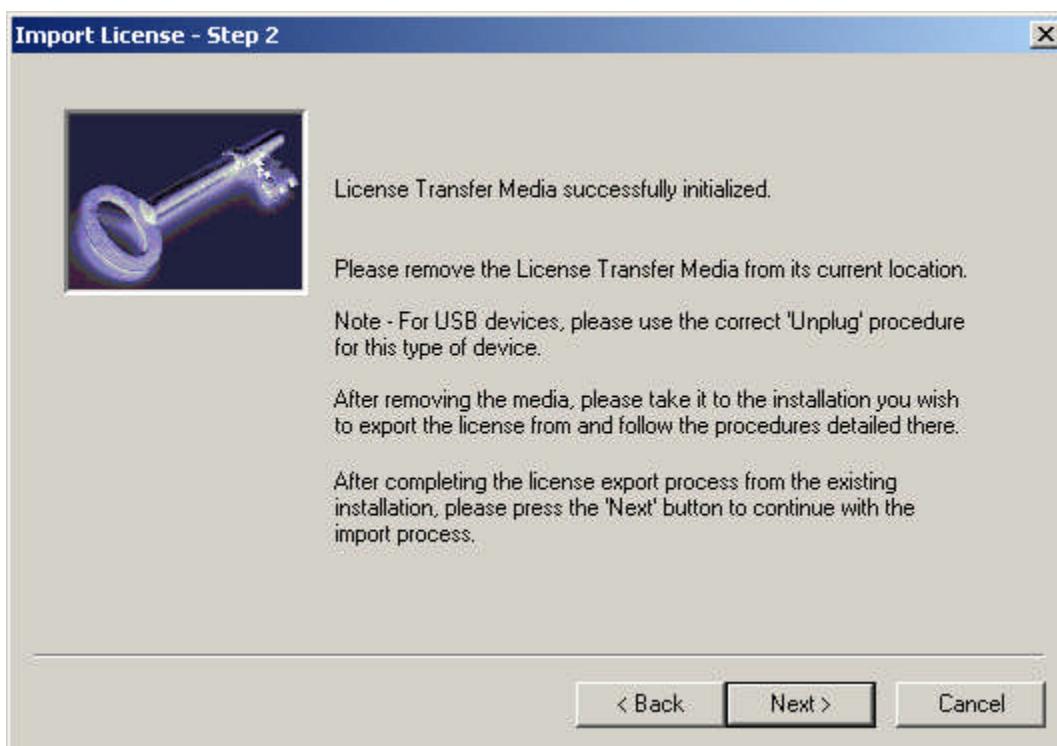
When the 'Finish' button is pressed, the application will terminate. This completes the license export.

Remove and take the 'License Transfer Media' to the newly installed WinEyeQ.

Note: For USB devices please follow the correct unplug procedure for your device.

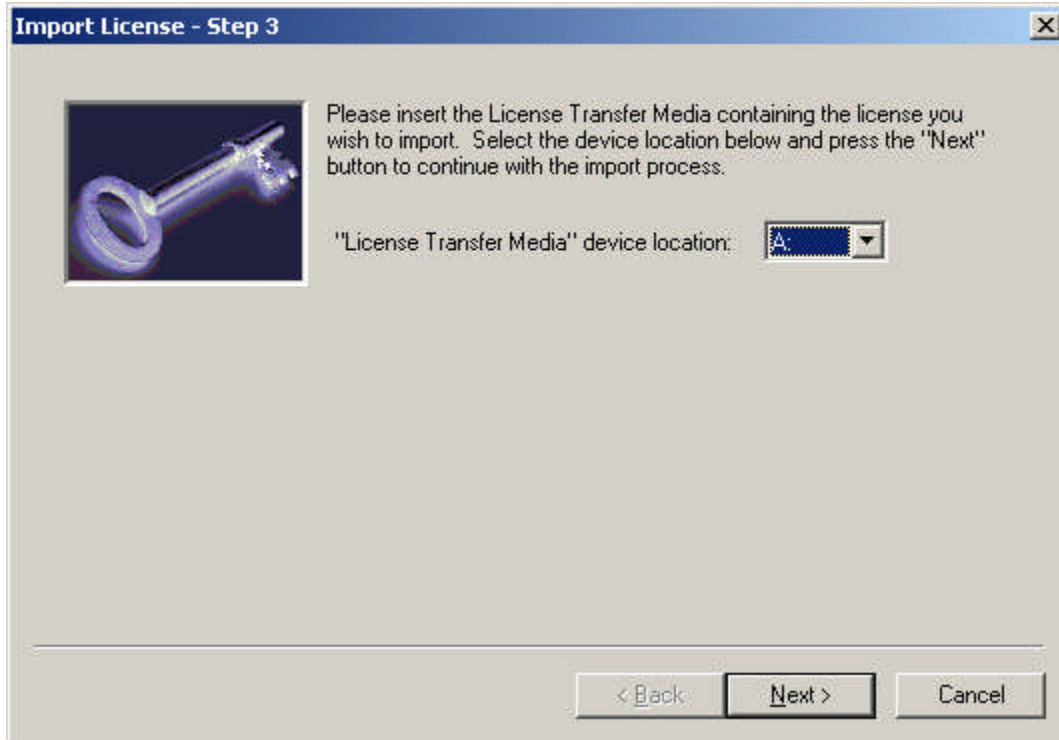
Step Three - Install Exported License

The PC with the newly installed version of WinEyeQ should still have the following screen displayed, 'Import License – Step 2':



After the export procedure is complete, and you have the license on the transfer media, insert or plug in the media and then press the 'Next' button.

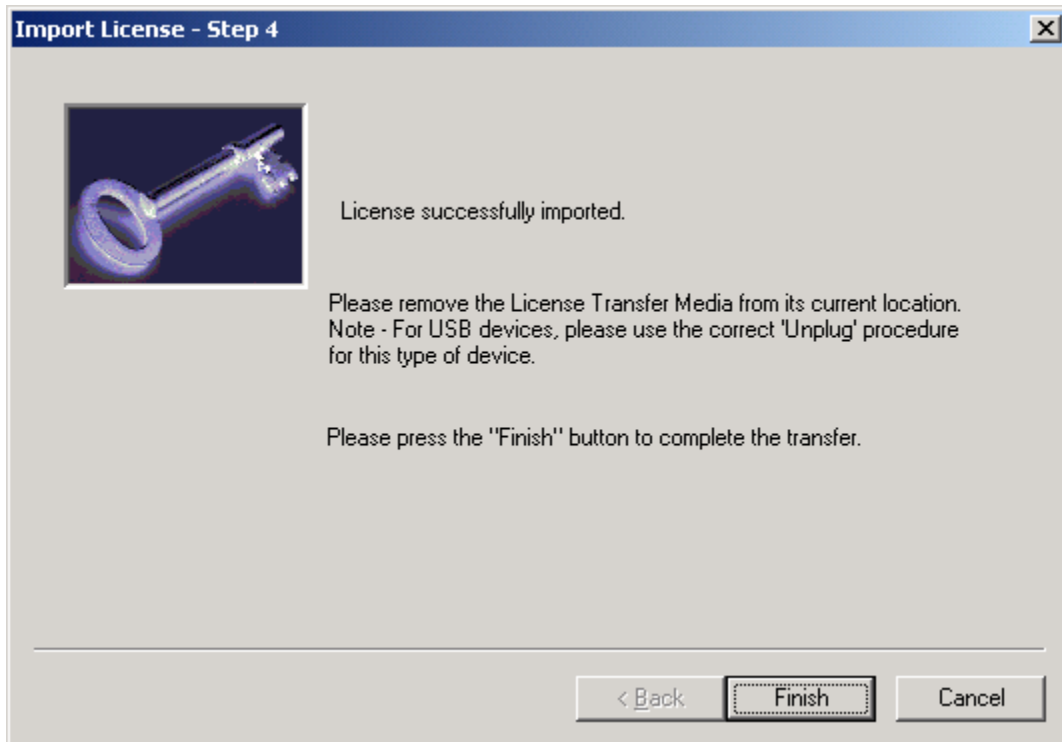
Select the proper 'License Transfer Media':



Press the 'Next' button when done.

WinEyeQ User's Guide

When the license has been successfully imported, the following dialog will appear:



The newly installed WinEyeQ is now fully enabled and ready to run when you press the 'Finish' button.

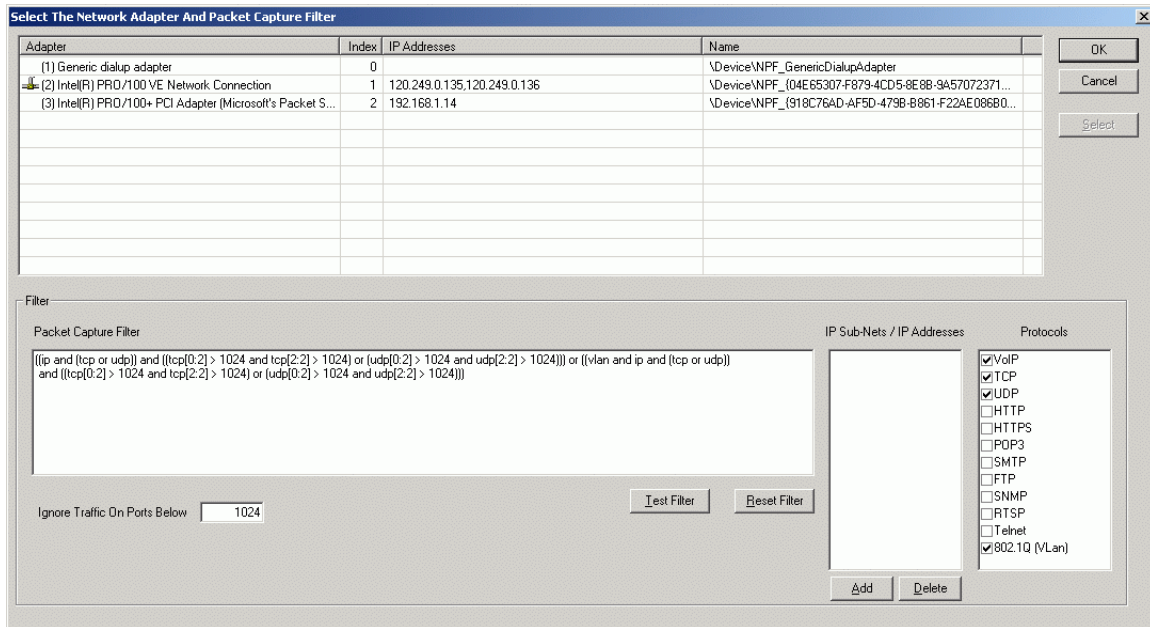
License Transfer Instruction Chart

Action	New Installation	Existing Installation
1.Install New Software	<p>Select the machine on which you would like to install the new instance of the product and follow the installation instructions.</p> <p>Once installed, run the application and the licensing dialog will appear.</p>	
2. Initialize License Transfer Media	<p>When the new installation asks for the Authorization code, press the 'Advanced' button then, press the 'Import License' button. This will bring up a dialog that asks you to initialize a 'License Transfer Media Device'. This device may be a diskette or USB device.</p> <p>Enter the letter of the drive where the transfer media is located and press the 'Next' button. Once you have pressed the 'Next' button, you may remove the License Transfer Device. You must then take that diskette or USB device to the PC that has the license you want to export.</p>	

Action	New Installation	Existing Installation
3. Export License		<p>Run the application on the PC that has the license you want to export, go to the Help menu and press Licensing Information.</p> <p>Press the 'Advanced' button to reveal the advanced options. Once visible, press the 'Export License' button.</p> <p>Follow the step-by-step directions to export the license onto the License Transfer Media Device.</p> <p>Remove the License Transfer Media Device. The existing installation is now deactivated.</p> <p>Return to the new installation.</p>
4. Import License	<p>Insert your License Transfer Device into the appropriate device. Follow the instructions to import the license. The new installation is now activated.</p>	

Selecting the Network Adapter

One of the first steps in preparing to run WinEyeQ is to select the network adapter you wish to monitor. WinEyeQ will automatically display the Select Adapter screen immediately after starting it for the first time. You may also access this dialog from the Edit | Select Adapter menu item.



On the top part of the screen is a list of the Network Adapters that WinEyeQ has discovered on your PC. Select the adapter you want to monitor by clicking the adapter line and then pressing 'Select' or by just double clicking the adapter line.

This window will be discussed in great detail later in the manual.

WinEyeQ User Interface

WinEyeQ was designed to facilitate diagnostics by representing the network in a natural, intuitive, top-down manner. This presentation allows users to “drill-down” into areas of interest at the same time bypassing information that is neither relevant nor interesting at the moment.

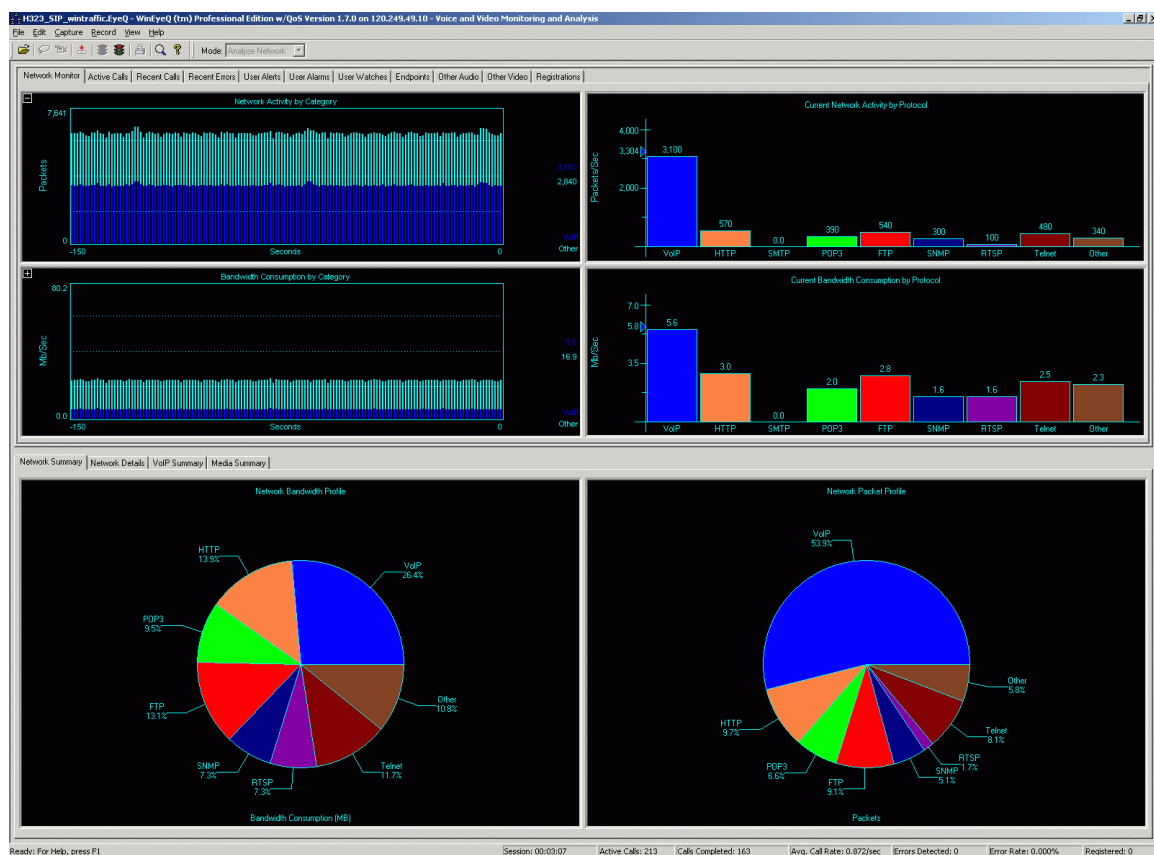
Data Scopes™

Version 1.5.0 of WinEyeQ implemented a new series of graphical representations of both the Voice and Video over IP and non-VoIP components of your network. The “Data Scope” metaphor reinforces WinEyeQ’s drill-down user-interface approach. Each Data Scope™ is represented at its topmost level by a view of logically grouped components (e.g. network protocols) in a view that

WinEyeQ User's Guide

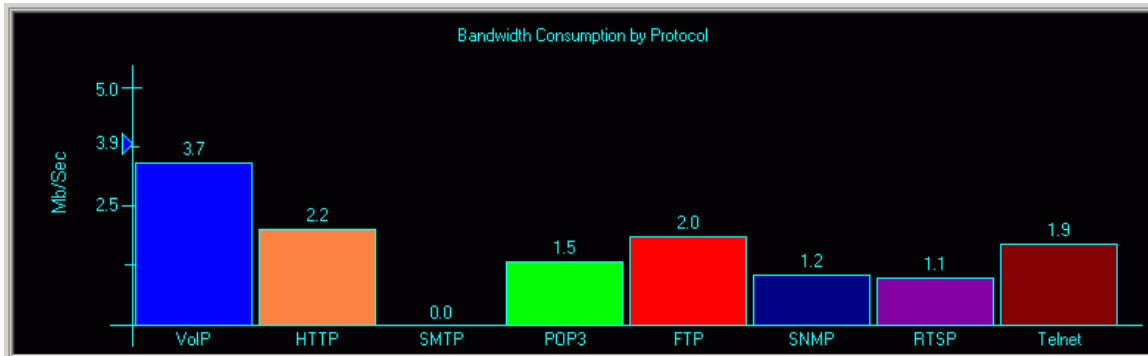
can be toggled between a pie chart and a bar chart. Each of these components has at least one level of depth beyond the first, which minimally would be a historical representation of the values of the component over time, which we refer to as a “histogram”. At its most complex, the topmost Data Scope™ will be the highest representation of a series of cascading views which each end at a histogram. The following gives you an overview of the typical mechanism of a Data Scope™ for isolating the G.723 bandwidth utilization on a live VoIP network.

Network Monitor View



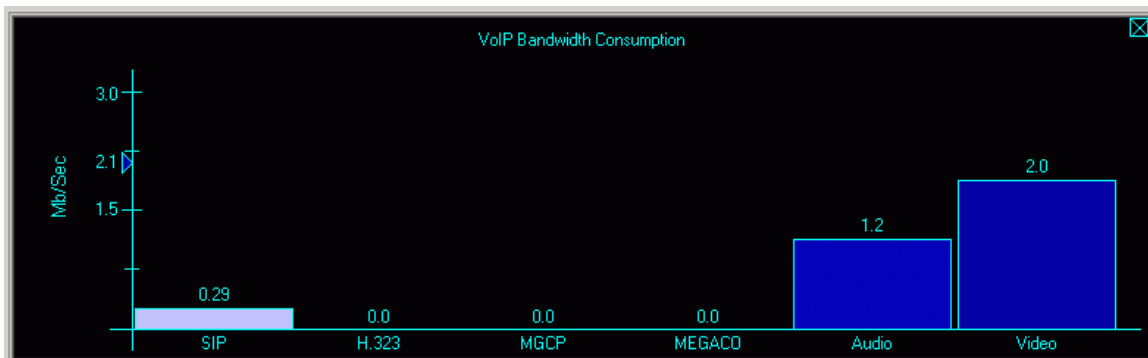
If we zero in on the Network Bandwidth Data Scope™ (found in the bottom of the upper right quadrant), we see a Data Scope™ that appears as follows:

Network Bandwidth Consumption; top view



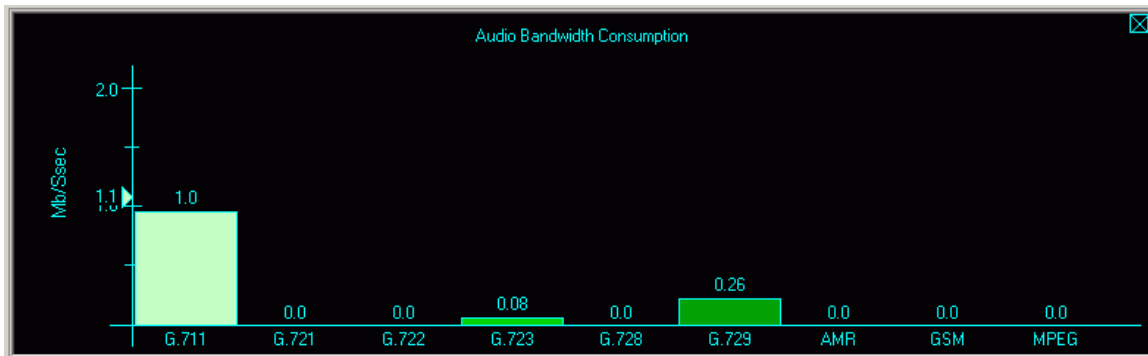
Notice that the components are grouped logically and that this Data Scope™ provides a high-level view of the bandwidth utilization of the various protocols on the network. The leftmost component is the VoIP component. To further explore the bandwidth utilization of the VoIP component, we can drill down by double-clicking on it. This action would yield a view of the VoIP breakdown as:

VoIP Bandwidth Consumption; level 2



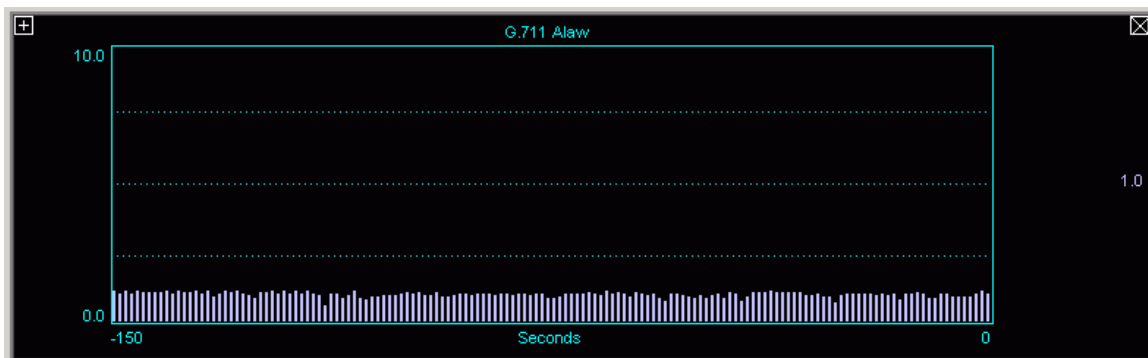
Once again, the components of this sub-level Data Scope™ are grouped logically, representing the top-level view of the bandwidth utilization of the VoIP components. The leftmost component is the SIP component, followed by H.323, Audio, and Video components. To further explore the bandwidth utilization of the Audio component, we can drill down by double-clicking on it. This action would yield a view of the Audio components as:

Audio Bandwidth Consumption; level 3



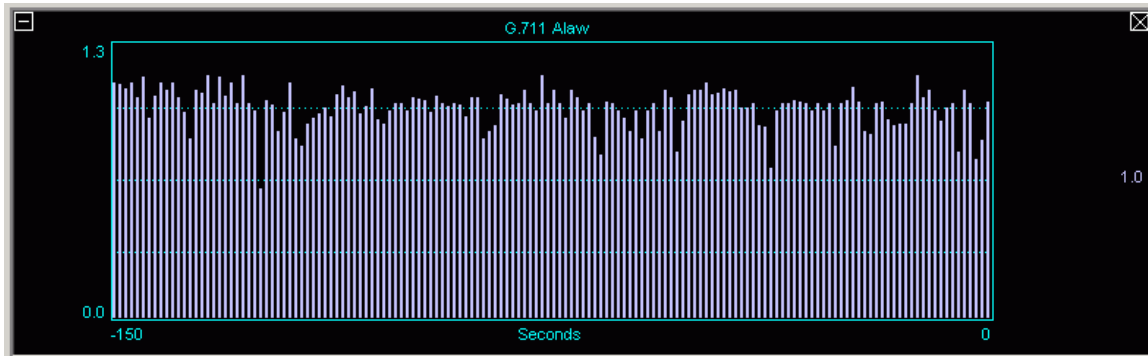
The components of this sub-level Data Scope™ are grouped logically representing the bandwidth utilization of the audio component by codec type. The leftmost component is the G.711 codec, which also has a sub-level Data Scope™ further refining it to the Alaw and Ulaw components. To further explore the bandwidth utilization of the G.723 component, we can drill down by double-clicking on it. This action would yield a view of the G.723 component as:

G.711 Bandwidth Consumption Histogram; level 4

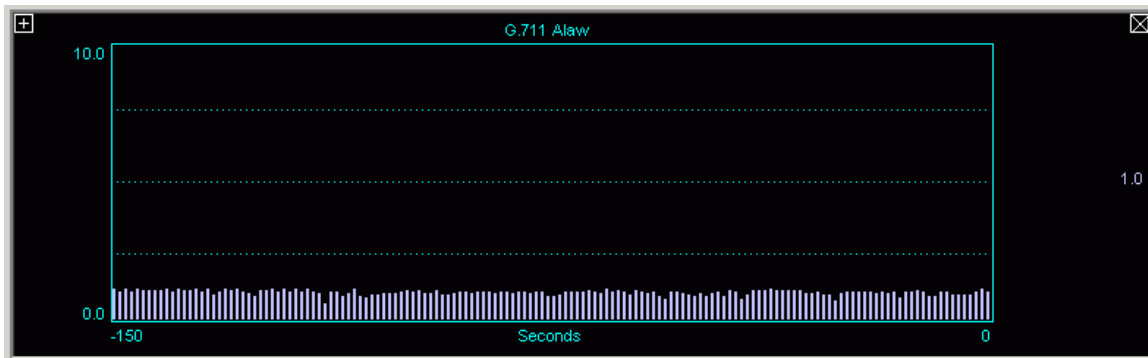


By clicking the “+” sign on the histogram, we can “Zoom In” on the series of values.

G.711 Bandwidth Consumption Histogram; zoomed



By clicking the “-” sign on the histogram, we can “Zoom Out” on the series of values back to:



Once you reach the histogram of a component you are at the end of the journey. You may back out from any sub-level at any time by using the “X” in the upper-right corner. The following section provides the user-interface tips and tricks for using the data scopes:

Data Scopes™ in Bar Graph View provide high-water marks for the component with the highest value on the scope. These marks, indicated by an arrow on the left scale, have the same color as the component that they are associated with. These watermarks are re-calculated every 10 updates of the Data Scope.

Navigational Tips

- Right-click the background area of a Data Scope™ to toggle between Bar Graph View and Pie Chart View.
- Double-click components to drill-down.
- Click the “X” box on a sub-level component to navigate backwards.
- Right-click any component to view its histogram.
- Click the “+” box to zoom-in the scale on a histogram.
- Click the “-” box to zoom-out the scale on a histogram.

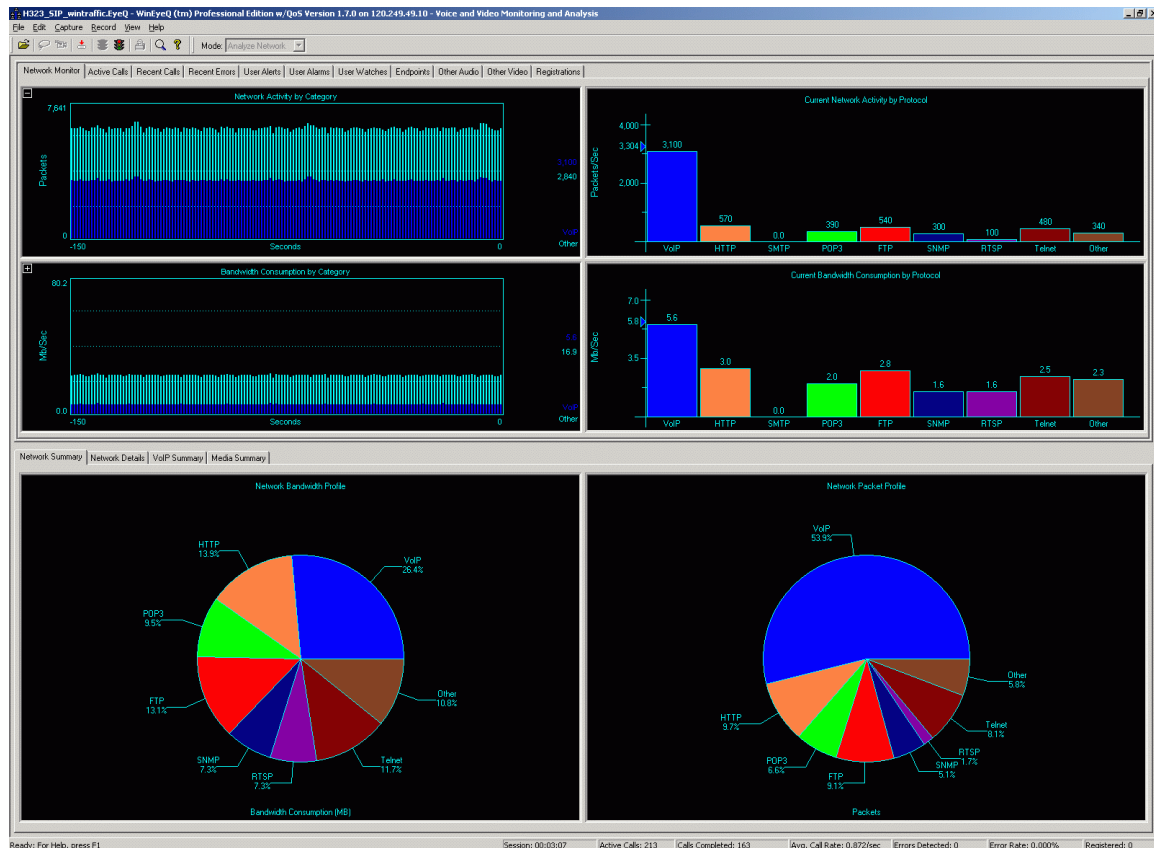
User Interface: Step-By-Step

WinEyeQ’s user interface is comprised of eleven (11) major views each containing up to eleven (11) sub-views. The ten major views represent the following categories:

- Network Monitor
- Active Call
- Recent Calls
- Recent Errors
- User Alerts
- User Alarms
- User Watches
- Endpoints
- Other Audio Channels
- Other Video Channels
- Registrations

The Network Monitor View

For the main view (Network Monitor) the Data Scopes™ are paired in Activity/Bandwidth pairs for logical groups of components. For example, Network Protocol Activity and Network Bandwidth Consumption by Protocol are paired together.

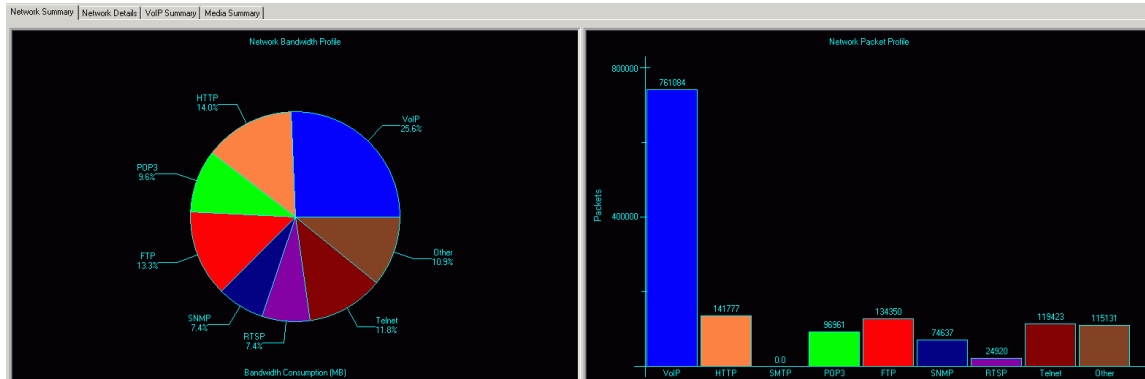


The Network Monitor View is designed to provide an overall picture of VoIP and Non-VoIP network traffic and resource utilization both instantaneously (top half) and historically (bottom half). You may elect to “drill into” any of the component elements as demonstrated earlier. The network monitor view has the following sub-views:

- Network Summary
- Network Details
- VoIP Summary
- Media Summary

Network Summary

This view presents the high-level Data Scopes™. The graphs represent the Network Bandwidth and Packet Profiles by component over the duration of the session. In this example, the Network Packet Profile Data Scope™ is in Bar Chart Mode while the Network Bandwidth Summary is in Pie Chart Mode. These modes can be toggled back and forth by right clicking on the background of the Data Scope™.



Network Details

Network Summary Network Details VoIP Summary Media Summary			
Protocol	Packets	Bytes	
IP	16,372,645	7,327,712,016	
ICMP	13,497	1,180,157	
UDP	15,888,954	6,973,748,744	
TCP	550,170	25,330,127	
H.323	321,574	12,124,950	
RAS	0	0	
TPKT	308,958	13,411,246	
H.225	58,778	4,584,614	
H.245	262,796	7,540,336	
SIP	102,146	54,863,916	
MGCP	0	0	
MEGACO	0	0	
RTP	15,616,208	6,774,823,538	
RTCP	90,010	17,506,208	
HTTP	0	0	
HTTPS	0	0	
SMTP	0	0	
POP3	0	0	
FTP	0	0	
SNMP	0	0	
RTSP	0	0	
Telnet	0	0	
Other	3,176	506,024	

Processing			Packets		(%)
TotalPacketsReceived			16,372,667		
PacketsProcessed			16,372,664		100.00
PacketsMissed			0		0.00
PacketsDiscarded			0		0.00
TotalRTPPacketsLost			0		0.00
Packets/Sec. (Avg)			5,220.88		

Call Metrics		Value
Current SIP Calls		29
Total SIP Calls Passed		14,572
Total SIP Calls Failed		0
Current H.323 Calls		24
Total H.323 Calls Passed		14,670
Total H.323 Calls Failed		0
Current Calls With Audio		49
Current Calls With Video		24
Current Endpoints		102

Average Network Metrics		Value
Audio Jitter (ms)		1.20
Audio Listening MOS (% of Optimal)		94.77
Audio Listening R Factor (% of Optimal)		93.53
Audio Conversational MOS (% of Optimal)		94.26
Audio Conversational R Factor (% of Optimal)		92.77
Video Jitter (ms)		1.43
Media Jitter (ms)		1.28
Initial Response Time (ms)		0.0383

Maximums		Value
VoIP Calls		77
VoIP Bandwidth(Mb/s)		37.22
VoIP Packets/Sec		11,057.83

This view provides a numerical summary of the packets and byte counts analyzed by layer. The layers include:

- IP, ICMP, UDP, TCP
- H.323, RAS, TPKT, H.225, H.245
- SIP
- MGCP, MEGACO
- RTP, RTCP
- HTTP, HTTPS, SMTP, POP3, FTP, SNMP, RTSP, Telnet, Other

Additional network metrics include:

Processing:

Total Packets Received: The total number of packets that WinEyeQ has received from the WinPcap driver.

Packets Processed: The number of packets that WinEyeQ has processed and analyzed.

Packets Missed: The number of packets that the WinPcap driver has been unable to send to WinEyeQ.

Packets Discarded: The number of packets that WinEyeQ has discarded due to packet overload.

Total RTP Packet Lost: The total number of RTP packets that were expected minus the total number actually received.

Packets per Second (Average): The average number of packets per second that WinEyeQ has processed since the analyzer was started.

Call Metrics:

Current SIP Calls

Total SIP Calls Passed

Total SIP Calls Failed

Current H.323 Calls

Total H.323 Calls Passed

Total H.323 Calls Failed

Current Calls with Audio

Current Calls with Video

Current Endpoints

Average Network Metrics:

Audio Jitter (ms): The average jitter (as calculated from RFC 3550) for all audio streams of all completed calls.

Audio Listening MOS (% of Optimal): The average Listening MOS score attained for all audio streams of all completed calls. See below.

Audio Listening R Factor (% of Optimal): The average Listening R factor attained for all audio streams of all completed calls. See below.

Audio Conversational MOS (% of Optimal): The average Conversational MOS score attained for all audio streams of all completed calls. See below.

Audio Conversational R Factor (% of Optimal): The average Conversational R factor attained for all audio streams of all completed calls. See below.

Video Jitter (ms): The average jitter (as calculated from RFC 3550) for all video streams of all completed calls.

Media Jitter (ms): The combined average of the audio and video jitter values.

Initial Response Time (ms): The average time it took for the called endpoint to return its first response to the calling endpoint.

Maximums:

VoIP Calls: The maximum number of concurrent calls that WinEyeQ has analyzed.

VoIP Bandwidth (Mb/s): The highest VoIP bandwidth analyzed.

VoIP Packets / Sec: The highest number of VoIP packets per second analyzed.

Packets/Second: The highest number of packets per second analyzed.

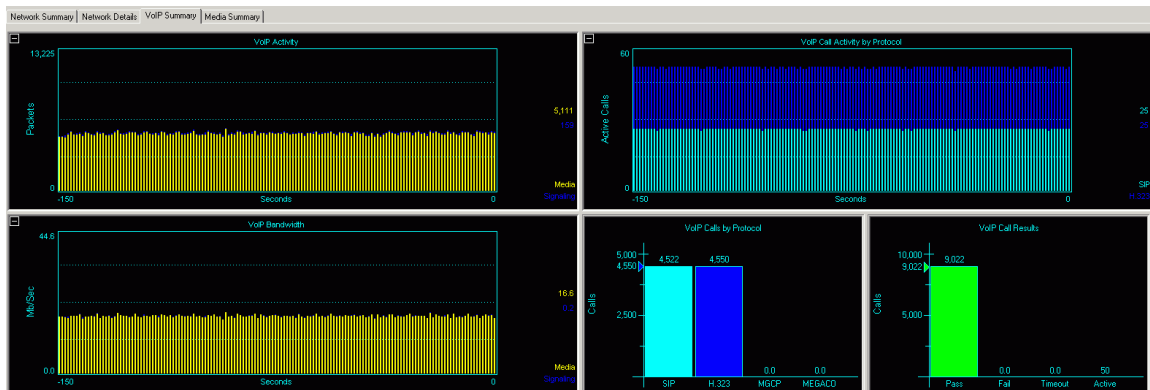
Optimal MOS Scores and R Factors.

Different codec types have different highest attainable Listening and Conversational MOS scores and R factors. WinEyeQ computes the normalized average network MOS scores and R factors by taking the MOS scores and R factors calculated for the audio media stream and dividing them by their theoretical maximum values. For example, if a G.728 audio stream received a Listening MOS score of 3.5, the normalized value would be 86.6 %. If a G.723.1 5.3 kb audio stream received a Listening MOS score of 3.5, the normalized value would be 96.9 %.

Note: Please see Appendix A for a chart of theoretical maximum MOS scores and R factors.

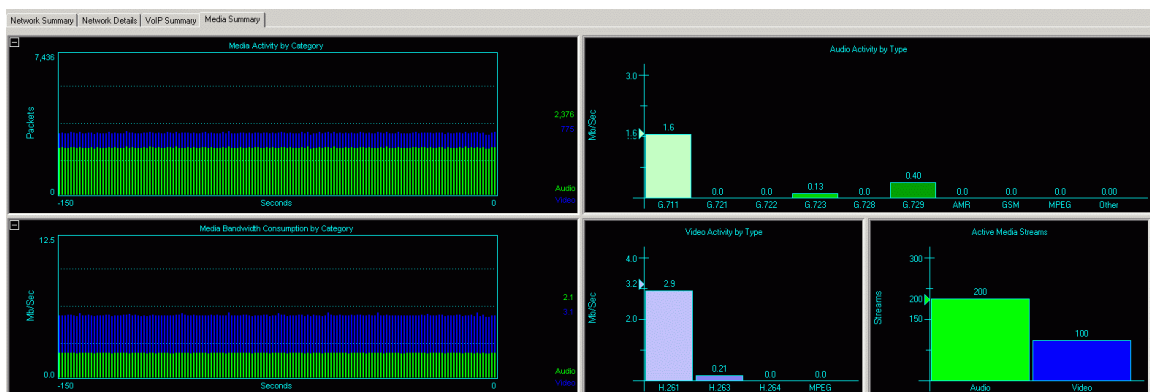
VoIP Summary

The VoIP Summary paints a picture of the packet and bandwidth activity of the VoIP signaling and media components as well as detailing call activity by protocol, call distribution by protocol, and call status history.



Media Summary

The Media Summary shows the bandwidth consumption and activity of the media components of the VoIP activity on your network. The bandwidth and packet activity are broken down by audio and video components. The right half of the screen breaks audio and video down by codec type as well as summarizing the active audio and video streams on the network.



Active Calls View

Active Calls View Summary Table:

Signaling	Value	Audio	Value	Video	Value
Src Address	120.249.1.13	Src Audio Channel	120.249.1.13:50616	Src Video Channel	120.249.1.13:50518
Src E.164	10002	Src Media Type	G.729	Src Media Type	H.263
Src H.323 ID	20002	Src Packet Count	227	Src Packet Count	208
Dest Address	120.249.1.11	Src Average Jitter (ms)	0.727	Src Average Jitter (ms)	0.967
Dest E.164	50002	Src Average Packet Interval (ms)	60.949	Src Average Packet Interval (ms)	66.774
Dest H.323 ID	60002	Src Average Bandwidth (kb/s)	6.034	Src Average Bandwidth (kb/s)	0.000
Start Time	12:04:41	Src Packets Lost	0	Src Packets Lost	0
Stop Time		Src TOS/DSCP Flag	Default (000000)	Src TOS/DSCP Flag	Default (000000)
Duration		Src Listening R Factor	83		
		Src Listening MOS Score	3.95		
		Optimal Listening R Factor(MOS Score)	83(5.95)		
		Stream Quality Index (SQI)	A+ (100.00)		
Call Terminator		Dest Audio Channel	120.249.1.11:50716	Dest Video Channel	120.249.1.11:50718
Gatekeeper		Dest Media Type	G.723 5.3k	Dest Media Type	H.263
Recording	No	Dest Packet Count	153	Dest Packet Count	208
Recorded	No	Dest Average Jitter (ms)	1.089	Dest Average Jitter (ms)	0.810
Captured	No	Dest Average Packet Interval (ms)	90.709	Dest Average Packet Interval (ms)	66.362
Record Filename		Dest Average Bandwidth (kb/s)	5.452	Dest Average Bandwidth (kb/s)	0.000
Capture Filename		Dest Packets Lost	0	Dest Packets Lost	0
		Dest TOS/DSCP Flag	Default (000000)	Dest TOS/DSCP Flag	Default (000000)
		Dest Listening R Factor	74		
		Dest Listening MOS Score	3.61		
		Optimal Listening R Factor(MOS Score)	74(3.61)		
		Stream Quality Index (SQI)	A+ (100.00)		

Ready: For Help, press F1 Session: 00:02:13 Active Calls: 162 Calls Completed: 112 Avg. Call Rate: 0.842/sec Errors Detected: 0 Error Rate: 0.000% Registered: 0

The active calls view is designed to provide an in-depth view of each VoIP call and its status. Each call is represented by an entry in the topmost report; the entries are updated once every second. This view contains the following columns:

Call Status: The current status of the call. These may be things such as connecting, ringing, connected, error, etc.

Protocol: The values for this field are SIP or H.323.

Started: This is the time (local time) that the call was started.

Duration: The length of time the call is (or was) active.

Terminator: Which side of the call (Source or Destination) terminated the call.

Source Address: The address of the call initiator (caller).

Source ID/E.164: The SIP user ID or H.323 E.164 alias of the caller.

Source Name/H.323 ID: The SIP display name or H.323 ID of the caller.

Destination Address: The address of the call receiver (party called).

Destination ID/E.164: The SIP user ID or H.323 E.164 alias of the party called.

Destination Name/H.323 ID: The SIP display name or H.323 ID of the party called.

Call ID: The SIP or H.323 call ID associated with this call.

Registered With: The gatekeeper's IP address for H.323 calls, or the Proxy's IP address for SIP calls.

Conference ID: The conference ID (H.323 calls only).

Each individual call has the following sub-views.

- Call Summary
- Call Flow (ladder diagram)
- Call Trace
- Call Metrics
- Audio Summary
- Audio Details
- Audio QoS
- Video Summary
- Video Details
- Data Details
- RTCP Summary
- RTCP XR Summary
- DTMF Summary

Note: To display information about a particular call, select it (click the call line) in the call list. Whenever a call is selected, it will remain "locked" in the view for as long as you wish to view its details.

Call Summary

Call Summary			Call Flow			Call Trace			Call Metrics			Audio Summary			Audio Details			Audio QoS			Video Summary			Video Details			Data Details			RTCP Summary			RTCP XR Summary			DTMF Summary		
Signaling			Value			Audio			Value			Video			Value			Value			Value			Value			Value			Value			Value			Value		
Src Address			120.249.1.13			Src Audio Channel			120.249.1.13/50516			Src Video Channel			120.249.1.13/50518			Src Address			120.249.1.13			Src Audio Channel			120.249.1.13/50516			Src Video Channel			120.249.1.13/50518					
Src E.164			10002			Src Media Type			G.729			Src Media Type			H.261			Src E.164			10002			Src Media Type			G.729			Src Media Type			H.261					
Src H.323 ID			20002			Src Packet Count			227			Src Packet Count			207			Src H.323 ID			20002			Src Packet Count			227			Src Packet Count			207					
						Src Average Jitter (ms)			0.727			Src Average Jitter (ms)			0.987									Src Average Jitter (ms)			0.727			Src Average Jitter (ms)			0.987					
Dest Address			120.249.1.11			Src Average Packet Interval (ms)			60.049			Src Average Packet Interval (ms)			66.774			Dest Address			120.249.1.11			Src Average Packet Interval (ms)			60.049			Src Average Packet Interval (ms)			66.774					
Dest E.164			50002			Src Average Bandwidth (kb/s)			8.034			Src Average Bandwidth (kb/s)			0.000			Dest E.164			50002			Src Average Bandwidth (kb/s)			8.034			Src Average Bandwidth (kb/s)			0.000					
Dest H.323 ID			60002			Src Packets Lost			0			Src Packets Lost			0			Dest H.323 ID			60002			Src Packets Lost			0			Src Packets Lost			0					
Start Time			12:04:41			Src TOS/DSCP Flag			Default (000000)			Src TOS/DSCP Flag			Default (000000)			Start Time			12:04:41			Src TOS/DSCP Flag			Default (000000)			Src TOS/DSCP Flag			Default (000000)					
Stop Time						Src Listening R Factor			63			Src Listening R Factor			3.96			Stop Time						Src Listening R Factor			63			Src Listening R Factor			3.96					
Duration						Optimal Listening R Factor/MOS Score			83/3.96			Optimal Listening R Factor/MOS Score			83/3.96			Duration						Optimal Listening R Factor/MOS Score			83/3.96			Optimal Listening R Factor/MOS Score			83/3.96					
Call Terminator						Stream Quality Index (SQI)			A+ (100.00)			Stream Quality Index (SQI)			A+ (100.00)			Call Terminator						Stream Quality Index (SQI)			A+ (100.00)			Stream Quality Index (SQI)			A+ (100.00)					
Gatekeeper						Dest Audio Channel			120.249.1.11/50716			Dest Audio Channel			120.249.1.11/50718			Gatekeeper						Dest Audio Channel			120.249.1.11/50716			Dest Audio Channel			120.249.1.11/50718					
						Dest Media Type			G.723 5.3k			Dest Media Type			H.263									Dest Media Type			G.723 5.3k			Dest Media Type			H.263					
						Dest Packet Count			153			Dest Packet Count			208									Dest Packet Count			153			Dest Packet Count			208					
Recording			No			Dest Average Jitter (ms)			1.089			Dest Average Jitter (ms)			0.810			Recording			No			Dest Average Jitter (ms)			1.089			Dest Average Jitter (ms)			0.810					
Recorded			No			Dest Average Packet Interval (ms)			90.709			Dest Average Packet Interval (ms)			66.362			Recorded			No			Dest Average Packet Interval (ms)			90.709			Dest Average Packet Interval (ms)			66.362					
Captured			No			Dest Average Bandwidth (kb/s)			5.423			Dest Average Bandwidth (kb/s)			0.000			Captured			No			Dest Average Bandwidth (kb/s)			5.423			Dest Average Bandwidth (kb/s)			0.000					
Record Filename						Dest Packets Lost			0			Dest Packets Lost			0			Record Filename						Dest Packets Lost			0			Dest Packets Lost			0					
Capture Filename						Dest TOS/DSCP Flag			Default (000000)			Dest TOS/DSCP Flag			Default (000000)			Capture Filename						Dest TOS/DSCP Flag			Default (000000)			Dest TOS/DSCP Flag			Default (000000)					
						Dest Listening R Factor			74			Dest Listening R Factor			3.41									Dest Listening R Factor			74			Dest Listening R Factor			3.41					
						Dest Listening MOS Score			74/3.61			Dest Listening MOS Score			74/3.61									Dest Listening MOS Score			74/3.61			Dest Listening MOS Score			74/3.61					
						Optimal Listening R Factor/MOS Score			A+ (100.00)			Optimal Listening R Factor/MOS Score			A+ (100.00)									Optimal Listening R Factor/MOS Score			A+ (100.00)			Optimal Listening R Factor/MOS Score			A+ (100.00)					
						Stream Quality Index (SQI)						Stream Quality Index (SQI)												Stream Quality Index (SQI)						Stream Quality Index (SQI)								

This sub-view provides a summary of the call elements including source and destination addresses for signaling and media. There are three panes on this sub-view.

Signaling Pane

Source Address: The IP address of the calling endpoint.

Source ID/Source E.164: The source ID (SIP) or E.164 (H.323) of the calling endpoint.

Source Name/Source H.323 ID: The source name (SIP) or H.323 ID (H.323) of the calling endpoint.

Destination Address: The IP address of the called endpoint.

Destination ID/E.164: The source ID (SIP) or E.164 (H.323) of the called endpoint.

Destination Name/H.323 ID: The source name (SIP) or H.323 ID (H.323) of the called endpoint.

Start Time: The time the first packet was seen on the network.

Stop Time: The time the last packet was seen on the network.

Duration: The difference between the start and stop time.

Call Terminator: The endpoint that terminated the call.

Proxy/Gatekeeper: The address of the proxy (SIP) or gatekeeper (H.323) that participated in the call.

Recording: Whether or not the call is presently being recorded.

Recorded: Whether or not the call was recorded.

Captured: Whether or not the call was captured.

Record Filename: The file name of the record file.

Capture Filename: The file name of the capture file.

Audio Pane

Source Audio Channel: The IP address and port of the calling endpoint.

Source Media Type: The type of codec being used to send the audio.

Source Packet Count: The number of packets sent on this channel.

Source Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Source Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

Source Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Source Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Source TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

Source Listening R Factor: The current Listening R factor for this media stream.

Source Listening MOS Score: The current Listening MOS Score for this media stream.

Optimal Listening R Factor / MOS Score: The highest scores that are attainable for this codec.

Stream Quality Index: The ratio of the current R Factor and MOS score to there optimal values.

Destination Audio Channel: The IP address and port of the called endpoint.

Destination Media Type: The type of codec being used to send the audio.

Destination Packet Count: The number of packets sent on this channel.

Destination Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Destination Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

Destination Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Destination Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Destination TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

Destination Listening R Factor: The current Listening R factor for this media stream.

Destination Listening MOS Score: The current Listening MOS Score for this media stream.

Optimal Listening R Factor / MOS Score: The highest scores that are attainable for this codec.

Stream Quality Index: The ratio of the current R Factor and MOS score to there optimal values.

Video Pane

Source Video Channel: The IP address and port of the calling endpoint.

Source Media Type: The type of codec being used to send the Video.

Source Packet Count: The number of packets sent on this channel.

Source Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Source Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

Source Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Source Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Source TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

Destination Video Channel: The IP address and port of the called endpoint.

Destination Media Type: The type of codec being used to send the Video.

Destination Packet Count: The number of packets sent on this channel.

Destination Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Destination Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

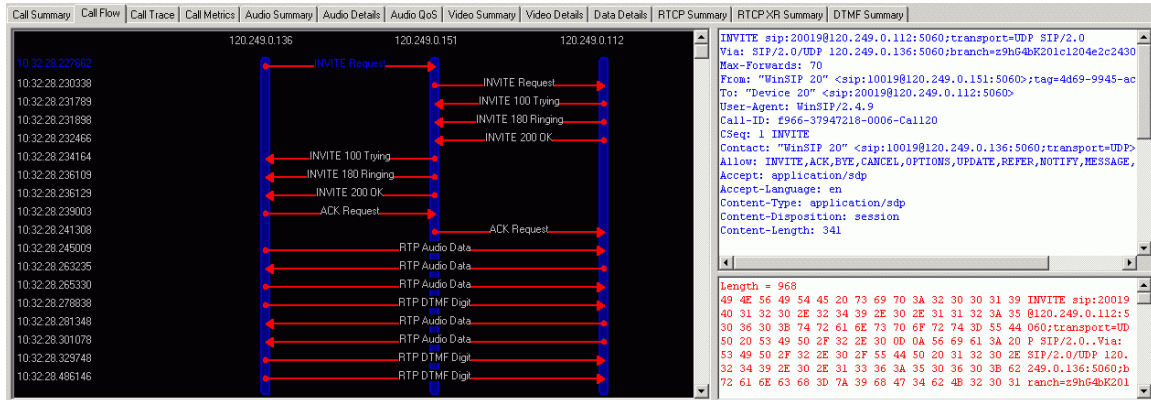
Destination Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Destination Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Destination TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

WinEyeQ User's Guide

Call Flow



This sub-view provides a time-stamped ladder diagram view of the call flow (signaling, media and media quality packets). Each “rung” in the ladder may be highlighted to display the decoded packet in both ASCII and hexadecimal representations.

Call Trace

Timestamp	Source IP	Port	Protocol	Method	Type	Code	Text	Dest IP	Port
10:38:34.448453	120.249.0.136	20008	SIP	INVITE	Request			120.249.0.151	5060
10:38:34.451944	120.249.0.151	5060	SIP	INVITE	Response	100	Trying	120.249.0.151	5060
10:38:34.452863	120.249.0.112	20010	SIP	INVITE	Response	180	Ringing	120.249.0.151	5060
10:38:34.452867	120.249.0.112	20010	SIP	INVITE	Response	200	OK	120.249.0.151	5060
10:38:34.452916	120.249.0.112	20010	SIP	INVITE	Response	200	OK	120.249.0.151	5060
10:38:34.454505	120.249.0.151	5060	SIP	INVITE	Response	100	Trying	120.249.0.136	5060
10:38:34.455682	120.249.0.151	5060	SIP	INVITE	Response	180	Ringing	120.249.0.136	5060
10:38:34.456774	120.249.0.151	5060	SIP	INVITE	Response	200	OK	120.249.0.136	5060
10:38:34.457845	120.249.0.136	20008	SIP	ACK	Request			120.249.0.151	5060
10:38:34.459666	120.249.0.151	5060	SIP	ACK	Request			120.249.0.112	5060
10:38:34.460622	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.465518	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.479560	120.249.0.112	40068	RTP					120.249.0.136	40068
10:38:34.484734	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.499419	120.249.0.112	40068	RTP					120.249.0.136	40068
10:38:34.518951	120.249.0.112	40068	RTP					120.249.0.136	40068
10:38:34.577471	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.733939	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.894987	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.945757	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.100960	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.151880	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.304035	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.354632	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.514801	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.565744	120.249.0.136	40068	DTM					120.249.0.112	40068

This sub-view provides a time stamped protocol specific display of the call flow (signaling, media and media quality packets). Each entry in the report may be highlighted to display the decoded packet in both ASCII and hexadecimal representations.

Call Metrics

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTP Summary RTPXR Summary DTMF Summary				
Metric	Value	Protocol	Packets	Bytes
Initial Response Time	00:00:00.003445	IP	4,032	762,257
Post-Dial Delay	00:00:00.003575			
Ring Duration	00:00:00.000231	ICMP	0	0
Time To Answer	00:00:00.003806	UDP	4,032	625,169
Time To Connect	00:00:00.006686	TCP	0	0
Teardown Time	00:00:00.000712			
Time Connected	00:01:00.0119703	H.323	0	0
End-To-End Time	00:01:00.026389	RAST	0	0
Signaling Latency	00:00:00.006495	TRT	0	0
Source Audio Delay	00:00:00.020227	H.225	0	0
Source Video Delay		H.245	0	0
Dest Audio Delay	00:00:00.011866	SIP	7	3,695
Dest Video Delay		RTP	4,001	588,172
		RTCP	24	1,056
		Other	0	0

This sub-view provides a summary of the call elements including metric measurements for response times and signaling interval and packets and byte counts analyzed by each protocol layer.

Initial Response Time: The length of time it took for the first message that was sent by the calling endpoint to be acknowledged by the called endpoint or proxy/gatekeeper.

Post-Dial Delay: The length of time from the start of the call until the start of the ring.

Ring Duration: The length of time the call was ringing.

Time to Answer: The length of time from the start of the call until it was answered.

Time to Connect: The length of time it took for the call to be connected.

Teardown Time: The length of time it took for the call close sequence to take place.

Time to Connect: The length of time from when the call was connected until the close sequence started.

End to End Time: The length of time from the start of the call until it was completed.

Signal Latency: The length of time it took for the call to connect plus disconnect.

Source Audio Delay: The length of time from when the call was connected until the first source audio packet was sent.

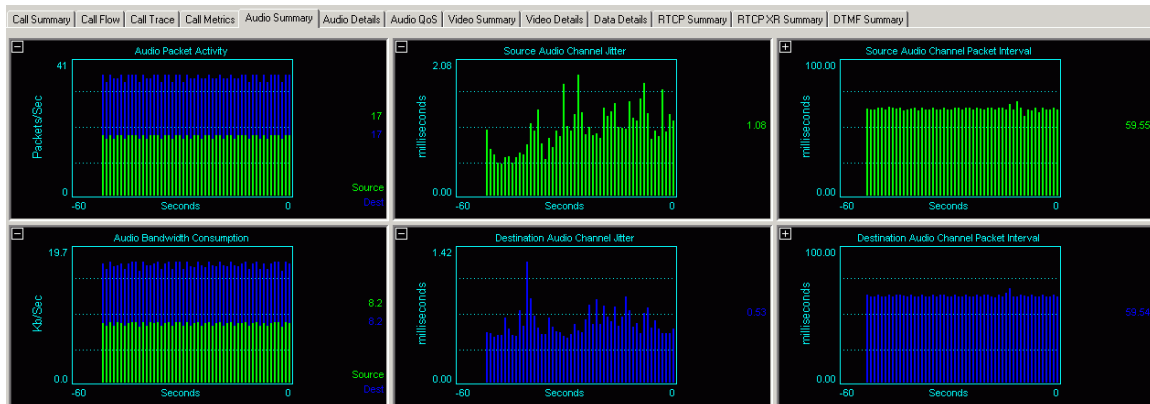
Source Video Delay: The length of time from when the call was connected until the first source video packet was sent.

Destination Audio Delay: The length of time from when the call was connected until the first destination audio packet was sent.

Destination Video Delay: The length of time from when the call was connected until the first destination video packet was sent.

Time to Admit (H.323): The length of time it took for the gatekeeper to acknowledge the ARQ message.

Audio Summary



This view provides jitter and latency measurements for the audio stream being sent by the calling and called parties. Included with the graphical representations of jitter and latency are the high low and current values for each as well as the stream type, the sender's IP address and port, the receiver's IP address and port, the number of packets lost and the DTMF sequences if present within the stream (RFC 2833 section 3 Named Telephony Events).

Audio Details

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTP Summary RTPXR Summary DTMF Summary					
Metric	Current	Low	High	Parameter	Source
Src Jitter (ms)	4.408	0.161	9.019	Address	120.249.1.13
occurred at:	00:00:01	00:00:04	00:00:04	Port	40252
Dest Jitter (ms)	3.169	0.782	5.977	Media Type	6.711 Allow
occurred at:	00:00:01	00:00:04	00:00:04	SSRC	28700ACD
Avg Src Packet Interval (ms)	20.124	2.137	77.041	Audio/Package (ms)	20
occurred at:	00:00:04	00:00:02	00:00:02	Frames/Package	20
Avg Dest Packet Interval (ms)	20.106	0.949	39.103	Total Packets	237
occurred at:	00:00:04	00:00:04	00:00:04	Packets Lost	0
Avg Src Bandwidth (kb/s)	64.761	64.286	66.667	Early Packets	76
occurred at:	00:00:05	00:00:01	00:00:01	Late Packets	58
Avg Dest Bandwidth (kb/s)	64.815	64.214	66.667	DTMF Events	62.49
occurred at:	00:00:05	00:00:01	00:00:01	Longest Packet Loss Burst	0
				Total Payload Bytes	37,920

This sub-view provides summary information including jitter and interval measurements for the audio streams. The high, low and current values for each stream as well as the stream type, the sender's IP address and port, the receiver's IP address and port, the number of packets lost and the DTMF sequences if present within the stream (RFC 2833 section 3 Named Telephony Events). This sub-view contains two panes.

Metrics Pane

Source Jitter (ms): The average, low, and high jitter measurements calculated for the source audio stream.

Occurred at:: The time, relative to the start of the call, that the high and low source jitter values were calculated.

Destination Jitter (ms): The average, low, and high jitter measurements calculated for the destination audio stream.

Occurred at:: The time, relative to the start of the call, that the high and low destination jitter values were calculated.

Average Source Packet Interval (ms): The average, low, and high inter-arrival time of packets on the source audio stream.

Occurred at:: The time, relative to the start of the call, that the high and low source packet interval values were calculated.

Average Destination Packet Interval (ms): The average, low, and high inter-arrival time of packets on the destination audio stream.

Occurred at:: The time, relative to the start of the call, that the high and low destination packet interval values were calculated.

Average Source Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the source audio stream.

Occurred at:: The time, relative to the start of the call, that the high and low source bandwidth values were calculated.

Average Destination Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the destination audio stream.

Occurred at:: The time, relative to the start of the call, that the high and low destination bandwidth values were calculated.

Parameters Pane

Address: The IP addresses for the source and destination channels.

Port: The port numbers for the source and destination channels.

Media Type: The codec type for the source and destination channels.

SSRC: The synchronization source for the source and destination channels.

Audio/Packet (ms): The length of audio time contained in each packet for the source and destination channels.

Frames/Packet: The number of audio frames contained in each packet for the source and destination channels.

Total Packets: The number of packets counted for the source and destination channels.

Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received for the source and destination channels.

WinEyeQ User's Guide

Early Packets: The number of packets considered early as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

Late Packets: The number of packets considered late as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

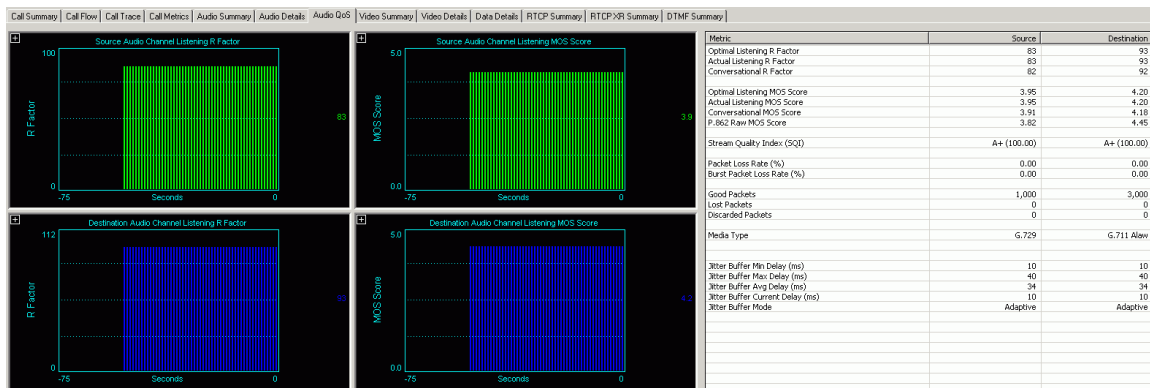
DTMF Events: The value of the DTMF digits (RFC 2833) for the source and destination channels.

Current Bandwidth (kb/s): The bandwidth, in kilobits per second, calculated during the last second for the source and destination channels.

Longest Packet Loss Burst: The count of the longest sequence of lost packets for the source and destination channels.

Total Payload Bytes: The number of bytes in the payload portion of the packet for the source and destination channels.

Audio QoS



This sub-view provides a real-time display of the R-factor and MOS scores for each stream. The R-factor/MOS scoring feature is a non-intrusive measurement technique available for the WinEyeQ call monitor. WinEyeQ passively measures the characteristics of live VoIP calls and reports quality scores in real-time. The algorithm used to obtain the R-Factor/MOS quality scores accurately models the way that time-varying impairments, most notably burst packet loss and possible jitter buffer discards, affect perceived speech quality. This sub-view has three panes.

Listening R Factor Pane

This pane displays the source audio (upper) and destination audio (lower) Listening R Factors in real-time.

Listening MOS Score Pane

This pane displays the source audio (upper) and destination audio (lower) Listening MOS Scores in real-time.

Metrics Pane

Optimal Listening R Factor: The highest score that is attainable for this codec.

Actual Listening R Factor: The current value of the Listening R Factor for the source and destination audio streams.

Conversational R Factor: The current value of the Conversational R Factor for the source and destination audio streams.

Optimal Listening MOS Score: The highest score that is attainable for this codec.

Actual Listening MOS Score: The current value of the Listening MOS Score for the source and destination audio streams.

Conversational MOS Score: The current value of the Conversational MOS Score for the source and destination audio streams.

P.862 Raw MOS Score: The current value of the P.862 Raw MOS Score for the source and destination audio streams.

Stream Quality Index: The ratio of the current R Factor and MOS score to their optimal values.

Packet Loss Rate (%): The total number packets that were lost divided by the total number of packets that were received.

Burst Packet Loss Rate (%): The packet loss rate encountered for burst conditions for the source and destination audio streams.

Good Packets: The number of packets received from the source and destination audio streams.

Lost Packets: The number of network lost packets from the source and destination audio streams.

Discarded Packets: The number of discarded packets due to excessive delay or extremely early arrival detected for the source and destination audio streams.

Media Type: The codec type for the source and destination audio streams.

Jitter Buffer Minimum Delay (ms): The minimum jitter buffer emulator delay in milliseconds occurring during a call for the source and destination audio streams.

Jitter Buffer Maximum Delay (ms): The maximum jitter buffer emulator delay in milliseconds occurring during a call for the source and destination audio streams.

Jitter Buffer Average Delay (ms): The average jitter buffer emulator delay in milliseconds occurring during a call from the source and destination audio streams.

WinEyeQ User's Guide

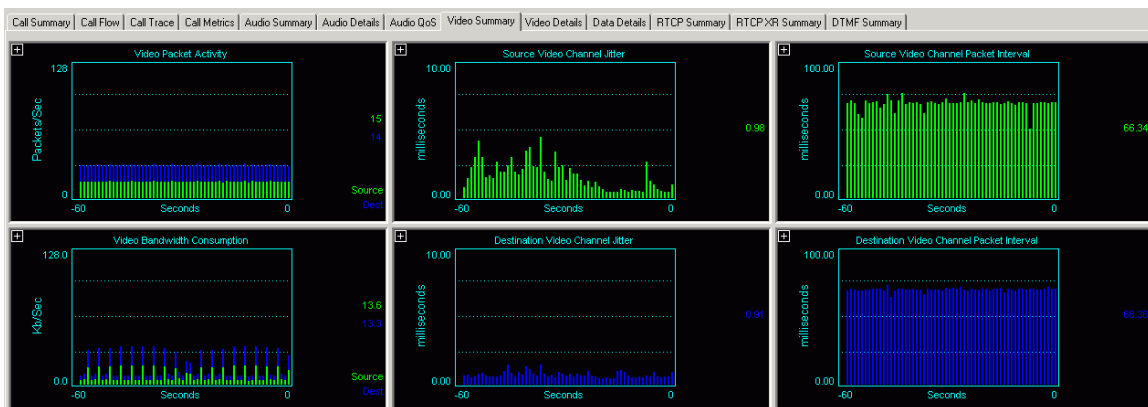
Jitter Buffer Current Delay (ms): The current jitter buffer emulator delay in milliseconds from the source and destination audio streams.

Jitter Buffer Mode: The type of jitter buffer (adaptive or fixer) being used for the source and destination audio streams. This is configured on the Edit Menu | Settings | QoS Tab.

The quality scores for MOS range from 0 to 4.5 and the R factor measurements range from 0 to 105 depending on codec type. The guidelines for interpreting the R-factor and MOS scores are shown in the table below for the G.711 codec:

Desirability Scale	R-factor Range	MOS Range
Desirable	94 - 80	4.4 - 4.0
Acceptable	80 - 70	4.0 - 3.6
Reach Connection	70 - 50	3.6 - 2.6
Not recommended	50 - 0	2.6 - 0

Video Summary



This view provides jitter and latency measurements for the video stream being sent by the calling and called parties. Included with the graphical representations of jitter and latency are the high low and current values for each stream, as well as the stream type, the sender's IP address and port, the receiver's IP address and port, the number of packets lost, the bandwidth consumption, the number of pictures detected, and the picture rate.

Video Details

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTP Summary RTP XR Summary DTMF Summary						
Metric	Current	Low	High	Parameter	Source	Destination
Src Jitter (ms)	0.768	0.054	2.804	Address	120.249.1.13	120.249.1.11
occurred at:		00:00:01	00:00:30	Port	50354	50554
Dest Jitter (ms)	0.923	0.028	2.028	Media Type	H.261	H.263
occurred at:		00:00:01	00:00:32	SSRC	D63BCD8	32736D1C
Avg Src Packet Interval (ms)	66.484	51.965	79.324	Total Packets	904	909
occurred at:		00:00:30	00:00:30	Packets Lost	0	0
Avg Dest Packet Interval (ms)	66.077	56.873	75.121	Early Packets	8	17
occurred at:		00:00:30	00:00:30	Late Packets	8	12
Avg Src Bandwidth (kb/s)	121.864	121.465	125.648	Pictures	904	909
occurred at:		00:00:56	00:00:01	Picture Rate	15.092	15.184
Avg Dest Bandwidth (kb/s)	8.855	8.438	11.939	Current Bandwidth (kb/s)	0.00	0.00
occurred at:		00:00:53	00:00:01	Longest Packet Loss Burst	0	0
				Total Payload Bytes	910,687	64,113

This sub-view provides summary information including jitter and interval measurements for the video streams. The measurements include high, low, and current values for each stream. As well as the stream type, the sender's IP address and port number, the receiver's IP address and port number, the packets lost, the number of pictures detected, and the picture rate. This sub-view contains two panes.

Metrics Pane

Source Jitter (ms): The average, low, and high jitter measurements calculated for the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source jitter values were calculated.

Destination Jitter (ms): The average, low, and high jitter measurements calculated for the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination jitter values were calculated.

Average Source Packet Interval (ms): The average, low, and high inter-arrival time of packets on the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source packet interval values were calculated.

Average Destination Packet Interval (ms): The average, low, and high inter-arrival time of packets on the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination packet interval values were calculated.

Average Source Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source bandwidth values were calculated.

Average Destination Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination bandwidth values were calculated.

Parameters Pane

Address: The IP addresses for the source and destination channels.

Port: The port numbers for the source and destination channels.

Media Type: The codec type for the source and destination channels.

SSRC: The synchronization source for the source and destination channels.

Total Packets: The number of packets counted for the source and destination channels.

Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received for the source and destination channels.

Early Packets: The number of packets considered early as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

Late Packets: The number of packets considered late as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

Pictures: The number of picture start codes counted for the source and destination channels.

Picture Rate: The number of pictures per second calculated for the source and destination channels.

Current Bandwidth (kb/s): The bandwidth, in kilobits per second, calculated during the last second for the source and destination channels.

Longest Packet Loss Burst: The count of the longest sequence of lost packets for the source and destination channels

Total Payload Bytes: The number of bytes in the payload portion of the packet for the source and destination channels.

Data Details

[illegible]

This sub-view provides summary information including interval measurements for the data streams, the high, low and current values for each stream as well as the stream type, the sender's IP address and port, the receiver's IP address and port, and the number of packets lost. There are two panes on this sub-view.

Metrics Pane

Average Source Packet Interval (ms): The average, low, and high inter-arrival time of packets on the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source packet interval values were calculated.

Average Destination Packet Interval (ms): The average, low, and high inter-arrival time of packets on the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination packet interval values were calculated.

Average Source Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source bandwidth values were calculated.

Average Destination Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination bandwidth values were calculated.

Parameters Pane

Address: The IP addresses for the source and destination channels.

Port: The port numbers for the source and destination channels.

Media Type: The codec type for the source and destination channels.

SSRC: The synchronization source for the source and destination channels.

Total Packets: The number of packets counted for the source and destination channels.

Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received for the source and destination channels.

Current Bandwidth (kb/s): The bandwidth, in kilobits per second, calculated during the last second for the source and destination channels.

Longest Packet Loss Burst: The count of the longest sequence of lost packets for the source and destination channels

Total Payload Bytes: The number of bytes in the payload portion of the packet for the source and destination channels.

RTCP Summary

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTCP Summary RTCP XR Summary DTMF Summary			
Source RTCP Channel Summary:			
Metric	Audio Channel	Video Channel	
Sender Address	120.249.49.11:29889	120.249.49.11:29891	
Receiver Address	120.249.0.150:26593	120.249.0.150:26595	
Sender Reports	1	1	
Receiver Reports	1	1	
SDES Reports	1	1	
Bye Reports	0	0	
Application Reports	0	0	
Senders RTP Packet Count	6	5	
Senders RTP Byte Count	2,880	1,601	
Reported Jitter (ms)	0.000	0.000	
Delay Since Last SR (sec)	0.00	0.00	
Reported Packets Lost	0	0	
Highest Sequence Number	0	0	
Fraction Lost (%)	0.00	0.00	
Canonical Name	Win323 Initiate Call 1	Win323 Initiate Call 1	
Name	Audio	Video	
E-Mail Address	support@touchstone-inc.com	support@touchstone-inc.com	
Phone Number	+215.672.6950	+215.672.6950	
Location	228 North York Road, Suite D, Hatboro, PA 19040 ...	228 North York Road, Suite D, Hatboro, PA 19040 ...	
Tool	Voice And Video Over IP Test Tool	Voice And Video Over IP Test Tool	
Note	Version 1.5.0	Version 1.5.0	
Private	www.touchstone-inc.com	www.touchstone-inc.com	
Destination RTCP Channel Summary:			
Metric	Audio Channel	Video Channel	
Sender Address	120.249.0.150:26593	120.249.0.150:26595	
Receiver Address	120.249.49.11:29889	120.249.49.11:29891	
Sender Reports	1	1	
Receiver Reports	1	1	
SDES Reports	1	1	
Bye Reports	0	0	
Application Reports	0	0	
Senders RTP Packet Count	37	123	
Senders RTP Byte Count	17,760	125,251	
Reported Jitter (ms)	0.000	0.000	
Delay Since Last SR (sec)	0.00	0.00	
Reported Packets Lost	0	0	
Highest Sequence Number	0	0	
Fraction Lost (%)	0.00	0.00	
Canonical Name	Win323 Terminate Call 31624	Win323 Terminate Call 31624	
Name	Audio	Video	
E-Mail Address	support@touchstone-inc.com	support@touchstone-inc.com	
Phone Number	+215.672.6950	+215.672.6950	
Location	228 North York Road, Suite D, Hatboro, PA 19040 ...	228 North York Road, Suite D, Hatboro, PA 19040 ...	
Tool	Voice And Video Over IP Test Tool	Voice And Video Over IP Test Tool	
Note	Version 1.5.3	Version 1.5.3	
Private	www.touchstone-inc.com	www.touchstone-inc.com	

This sub-view provides summary information that has been gathered from the RTCP packets that WinEyeQ has analyzed for the audio and video streams that have been sent by both endpoints of the call. This sub-view has two panes that are identical except for source and destination.

RTCP Channel Summary Pane

Sender Address: The IP address and port number of the sending RTCP channel.

Receiver Address: The IP address and port number of the receiving RTCP channel.

Sender Reports: The number of RTCP Sender Reports sent.

Receiver Reports: The number of RTCP Receiver Reports sent.

SDES: The number of RTCP SDES Reports sent.

Bye Reports: The number of RTCP Bye Reports sent.

Application Reports: The number of RTCP Application Reports sent.

Senders Packet Count: The total number of RTP data packets transmitted by the sender since starting transmission.

Senders Byte Count: The total number of payload octets transmitted in RTP data packets by the sender since starting transmission.

Reported Jitter (ms): The jitter measurement calculated on the stream being received from the other endpoint.

Delay Since Last SR (sec): The delay, expressed in units of 1/65536 seconds, between receiving the last SR packet from the remote endpoint and sending this reception report block.

Reported Packets Lost: The total number of RTP data packets from the remote endpoint that have been lost since the beginning of reception.

Highest Sequence Number: The low 16 bits contain the highest sequence number received in an RTP data packet from the remote endpoint, and the most significant 16 bits extend that sequence number with the corresponding count of sequence number cycles.

Fraction Lost (%): The fraction of RTP data packets from the remote endpoint lost since the previous SR or RR packet was sent, expressed as a fixed point number with the binary point at the left edge of the field.

Canonical Name: A unique end-point identifier.

Name: The real name used to describe the source.

E-Mail Address: The email address is formatted according to RFC 2822.

Phone Number: The phone number (should be formatted with the plus sign replacing the international access code).

Location: The geographic user location.

Tool: The application or tool name.

Note: This is intended for transient messages describing the current state of the source.

Private: This item is used to define experimental or application-specific extensions.

RTCP XR Summary

Cal Summary Cal Flow Cal Trace Cal Metrics Audio Summary Audio Di Audio QoS Video Summary Video Details Data Tables RTPCVR Summary DTMC Summary			
Source RTPCVR Channel Summary:			
Metric	Audio Channel	Video Channel	
Sender Address	120.249.49.10:25001	120.249.49.10:25003	
Receiver Address	120.249.0.150:25004	120.249.0.150:25043	
Extended Reports	22	22	
Loss Rate (%)	14.45	14.45	
Discard Rate (%)	0.00	0.00	
Average Burst Density (%)	19.53	19.53	
Average Gap Density (%)	0.00	0.00	
Average Burst Duration (ms)	1,444	1,444	
Average Gap Duration (ms)	445	445	
Round Trip Delay (ms)	16	16	
End System Delay (ms)	60	60	
Signal Level (db)	-105	-105	
Noise Level (db)	-67	-67	
Residual Echo Return Loss (db)	55	55	
Gap Threshold	16	16	
R Factor	90	90	
External R Factor	Unavailable	Unavailable	
Listening MOS	4.0	4.0	
Conversational MOS	4.0	4.0	
Packet Loss Concealment	Enhanced	Enhanced	
3ttr Buffer Adaptive	Adaptive	Adaptive	
3ttr Buffer Rate	8	8	
3ttr Buffer Nominal Delay (ms)	40	40	
3ttr Buffer Max Delay (ms)	40	40	
3ttr Buffer Absolute Max Delay (ms)	80	80	
Destination RTPCVR Channel Summary:			
Metric	Audio Channel	Video Channel	
Sender Address	120.249.0.150:25004	120.249.0.150:25043	
Receiver Address	120.249.49.10:25001	120.249.49.10:25003	
Extended Reports	22	22	
Loss Rate (%)	14.45	14.45	
Discard Rate (%)	0.00	0.00	
Average Burst Density (%)	19.53	19.53	
Average Gap Density (%)	0.00	0.00	
Average Burst Duration (ms)	1,444	1,444	
Average Gap Duration (ms)	445	445	
Round Trip Delay (ms)	16	16	
End System Delay (ms)	60	60	
Signal Level (db)	-105	-105	
Noise Level (db)	-67	-67	
Residual Echo Return Loss (db)	55	55	
Gap Threshold	16	16	
R Factor	90	90	
External R Factor	Unavailable	Unavailable	
Listening MOS	4.0	4.0	
Conversational MOS	4.0	4.0	
Packet Loss Concealment	Enhanced	Enhanced	
3ttr Buffer Adaptive	Adaptive	Adaptive	
3ttr Buffer Rate	8	8	
3ttr Buffer Nominal Delay (ms)	40	40	
3ttr Buffer Max Delay (ms)	40	40	
3ttr Buffer Absolute Max Delay (ms)	80	80	

This sub-view provides summary information that has been gathered from the RTCP XR packets that WinEyeQ has analyzed for the audio and video streams that have been sent by both endpoints of the call. This sub-tab has two panes that are identical.

RTCP XR Channel Summary

Sender Address: The IP address and port number of the sending RTCP channel.

Receiver Address: The IP address and port number of the receiving RTP channel.

Extended Reports: The number of RTCP Extended Reports sent.

Loss Rate (%): The fraction of packets lost since the beginning of the call.

Discard Rate (%): The fraction of packets discarded since the beginning of the call.

Average Burst Density (%): The fraction of packets within burst periods since the beginning of the call.

Average Gap Density (%): The fraction of packets within gap periods since the beginning of the call.

Average Burst Duration (ms): The mean duration, in milliseconds, of the burst periods since the beginning of the call.

Average Gap Duration (ms): The mean duration, in milliseconds, of the gap periods since the beginning of the call.

Round Trip Delay (ms): The most recently calculated round-trip delay, in milliseconds.

End System Delay (ms): The most recently estimated end system delay, in milliseconds.

WinEyeQ User's Guide

Signal Level (db): The relative speech signal level expressed as the ratio of the signal level to a 0 dBm0 reference.

Noise Level (db): The relative silence period noise level expressed as the ratio of the background noise level to a 0 dBm0 reference.

Residual Echo Return Loss (db): The residual echo return loss as the sum of the measured echo return loss (ERL) and the echo return loss enhanced (ERLE) of the echo canceller, expressed in dB.

Gap Threshold: The gap threshold, in packets.

R Factor: The voice quality metric for the call channel as measured in the monitored network segment.

External R Factor: The voice quality metric for the call channel as measured in an external monitored network segment.

Listening MOS: The estimated mean opinion listening quality score for the call channel.

Conversational MOS: The estimated mean opinion conversational quality score for the call channel.

Packet Loss Concealment: The packet loss concealment capabilities.

Jitter Buffer Adaptive: Adaptive or non-adaptive.

Jitter Buffer Rate: This represents the implementation specific adjustment rate of a jitter buffer in adaptive mode.

Jitter Buffer Nominal Delay (ms): The current nominal jitter buffer delay, in milliseconds.

Jitter Buffer Max Delay (ms): The maximum jitter buffer delay, in milliseconds, recorded for the call.

Jitter Buffer Absolute Max Delay (ms): The absolute maximum delay, in milliseconds, the jitter buffer can ever introduce to the call channel packet stream.

DTMF Summary

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTP/RTCP Summary RTP/RTCP Summary DTMF Summary																			
Source Keypad Event Summary:					2156726550					Destination Keypad Event Summary:									
Source Audio DTMF Event Detail										Destination Audio DTMF Event Detail									
Time	Elapsed	Event	Digit	Power	Duration	End Bit	Marker	Seq Num	TimeSta	Time	Elapsed	Event	Digit	Power	Duration	End Bit	Marker	Seq Num	TimeSta
09:40:09.352359	00:00:00.000000	dt02	2	0	1500	1	1	2	3										
09:40:09.402681	00:00:00.050322	dt02	2	0	1500	1	0	5	3										
09:40:09.453417	00:00:00.050736	dt01	1	0	1500	0	1	9	24										
09:40:09.554669	00:00:00.101551	dt01	1	0	1500	1	0	15	26										
09:40:09.605753	00:00:00.050785	dt01	1	0	1500	1	0	18	26										
09:40:09.656547	00:00:00.050794	dt05	5	0	1500	0	1	22	51										
09:40:09.758148	00:00:00.101601	dt05	5	0	1500	1	0	20	51										
09:40:09.809144	00:00:00.050996	dt05	5	0	1500	1	0	32	51										
09:40:09.859664	00:00:00.050520	dt06	6	0	1500	0	1	35	96										
09:40:09.961900	00:00:00.102236	dt06	6	0	1500	1	0	41	96										
09:40:10.012954	00:00:00.051054	dt06	6	0	1500	1	0	45	96										
09:40:10.063811	00:00:00.050957	dt07	7	0	1500	0	1	48	111										
09:40:10.166335	00:00:00.102524	dt07	7	0	1500	1	0	54	111										
09:40:10.218419	00:00:00.052004	dt07	7	0	1500	1	0	58	111										
09:40:10.270724	00:00:00.052305	dt02	2	0	1500	0	1	62	151										
09:40:10.372668	00:00:00.101944	dt02	2	0	1500	1	0	68	151										
09:40:10.423123	00:00:00.050455	dt02	2	0	1500	1	0	71	151										
09:40:10.473890	00:00:00.050767	dt06	6	0	1500	0	1	75	152										
09:40:10.575460	00:00:00.101570	dt06	6	0	1500	1	0	81	182										
09:40:10.626251	00:00:00.050791	dt06	6	0	1500	1	0	84	182										
09:40:10.677047	00:00:00.050796	dt05	5	0	1500	0	1	88	213										
09:40:10.778611	00:00:00.101564	dt05	5	0	1500	1	0	94	213										
09:40:10.829254	00:00:00.050743	dt05	5	0	1500	1	0	90	214										
09:40:10.880137	00:00:00.050703	dt05	5	0	1500	0	1	101	244										
09:40:10.981706	00:00:00.101569	dt05	5	0	1500	1	0	107	244										
09:40:11.032478	00:00:00.050772	dt05	5	0	1500	1	0	111	244										
09:40:11.083268	00:00:00.050790	dt00	0	0	1500	0	1	114	275										
09:40:11.184827	00:00:00.101559	dt00	0	0	1500	1	0	120	275										
09:40:11.255600	00:00:00.050773	dt00	0	0	1500	1	0	124	275										

This sub-view provides a detailed and organized tabular display for the active DTMF transmissions that occur during a call for both the source and destination side of the calls.

WinEyeQ User's Guide

Recent Calls View

WinEyeQ (tm) Professional Edition w/QoS Version 1.7.0 on 120.249.49.10, 120.249.49.11 - Voice and Video Monitoring and Analysis

File Edit Capture Record View Help

Mode Analyze Network

Network Monitor Active Calls Recent Calls Recent Errors User Alerts User Alarms User Watches Endpoints Other Audio Other Video Registrations

Status	Protocol	Started	Duration	Terminator	Source Address	Source ID/E.164	Source Name/H.323 ID	Destination Address	Destination ID/E.164	Destination Name/H.323 ID	Call ID	Registered With
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1019		120.249.49.21	2019		37C2932C4EAD9FF1AE9B07110CE57A	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1018		120.249.49.21	2018		ADA5A5C8014D4819EA413256FF712E994	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1017		120.249.49.21	2017		A95C3F8DAAD0A8A5B8E08F8FC5D63238	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1016		120.249.49.21	2016		ED4D6A7990F5D7641C76A9F0FF60542	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1015		120.249.49.21	2015		6214DC5F8FA0C875DE834163B48B0	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1014		120.249.49.21	2014		0F8BB1AF8B31109E9C6CD0940000E0E08	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1013		120.249.49.21	2013		345H56A7867485044CDBAFCDD4EACB	
✓ Released	H.323	09:28:14	00:01:00	Source	120.249.49.11	1012		120.249.49.21	2012		34687C5C70566F716F72EEC0691ACA	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1011		120.249.49.21	2011		59F8C694D0158F426208A51268E5218	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1009		120.249.49.21	2009		7A76DA200194C45E44A8742AEAC685C1	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1008		120.249.49.21	2010		0F81138F30F78F8F22D5B4A8F16F1	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1008		120.249.49.21	2008		C23645E38A3F91687E9E8F2C6A9FB	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1007		120.249.49.21	2007		7E830DE4C12C41688B4A2208EF49DA	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1006		120.249.49.21	2006		E4442F363D31F56E8EC4D83C0C5293D	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1005		120.249.49.21	2005		01656255942E788AA0CE3D0D4E49A	
✓ Released	H.323	09:28:13	00:01:00	Source	120.249.49.11	1004		120.249.49.21	2004		60DE8B6034132812490C6318E8BD	
✓ Released	H.323	09:28:12	00:01:01	Source	120.249.49.11	1003		120.249.49.21	2003		D08D05C2852325748E584DE2994D01	
✓ Released	H.323	09:28:12	00:01:00	Source	120.249.49.11	1002		120.249.49.21	2002		E2D29633F70ED94336F05D39095C05	
✓ Released	H.323	09:28:12	00:01:00	Source	120.249.49.11	1001		120.249.49.21	2001		AE245D08E063A9144EE4FC6CECDE8307D	
✓ Completed	SIP	09:27:37	00:01:00	Source	120.249.49.10	10019	"WinSIP 20"	120.249.49.20	20019	"Device 20"		
✓ Completed	SIP	09:27:37	00:01:00	Source	120.249.49.10	10018	"WinSIP 19"	120.249.49.20	20018	"Device 19"		
✓ Completed	SIP	09:27:37	00:01:00	Source	120.249.49.10	10017	"WinSIP 18"	120.249.49.20	20017	"Device 18"		
✓ Completed	SIP	09:27:37	00:01:00	Source	120.249.49.10	10016	"WinSIP 17"	120.249.49.20	20016	"Device 17"		
✓ Completed	SIP	09:27:37	00:01:00	Source	120.249.49.10	10015	"WinSIP 16"	120.249.49.20	20015	"Device 16"		
✓ Completed	SIP	09:27:37	00:01:00	Source	120.249.49.10	10014	"WinSIP 15"	120.249.49.20	20014	"Device 15"		

Call Summary | Call Flow | Call Trace | Call Metrics | Audio Details | Audio QoS | Video Details | Data Details | RTPCP Summary | RTPXR Summary | DTMF Summary

Signaling	Value	Audio	Value	Video	Value
Src Address	120.249.49.11	Src Audio Channel	120.249.49.11:25176	Src Video Channel	120.249.49.11:25178
Src E.164	1004	Src Media Type	G.729	Src Media Type	H.263
Src H.323 ID		Src Packet Count	1,000	Src Packet Count	909
Dest Address	120.249.49.21	Src Average Jitter (ms)	0.595	Src Average Jitter (ms)	0.575
Dest E.164	2004	Src Average Packet Interval (ms)	60.061	Src Average Packet Interval (ms)	66.074
Dest H.323 ID		Src Average Bandwidth (kb/s)	6.035	Src Average Bandwidth (kb/s)	8.857
Start Time	09:20:12	Src Packets Lost	0	Src Packets Lost	0
Stop Time	09:21:12	Src TOSS/OSCP Flag	Default (000000)	Src TOSS/OSCP Flag	Default (000000)
Duration	00:01:00	Src Listening R Factor	83		
Call Terminator	Source	Src Listening MOS Score	3.95		
Gatekeeper		Src Conversational R Factor	82		
Call Score	A+ 99.533	Src Conversational MOS Score	3.91		
Signaling Score	A+ 99.517	Stream Quality Index (SQI)	A+ (100.00)		
Media Score	A+ 99.507	Dest Audio Channel	120.249.49.21:25176	Dest Video Channel	120.249.49.21:25178
Src Aud Score	A+ 99.795	Dest Media Type	G.729	Dest Media Type	H.263
Dest Aud Score	A+ 99.552	Dest Packet Count	1,000	Dest Packet Count	909
Src Vid Score	A+ 99.508	Dest Average Jitter (ms)	1.815	Dest Average Jitter (ms)	0.879
Dest Vid Score	A+ 99.275	Dest Average Packet Interval (ms)	60.058	Dest Average Packet Interval (ms)	66.078
Record Filename		Dest Average Bandwidth (kb/s)	6.036	Dest Average Bandwidth (kb/s)	8.856
Capture Filename		Dest Packets Lost	0	Dest Packets Lost	0
		Dest TOSS/OSCP Flag	Default (000000)	Dest TOSS/OSCP Flag	Default (000000)
		Dest Listening R Factor	83		
		Dest Listening MOS Score	3.95		
		Dest Conversational R Factor	82		
		Dest Conversational MOS Score	3.91		
		Stream Quality Index (SQI)	A+ (100.00)		

Ready: For Help, press F1 Session: 00:09:22 Active Calls: 41 Calls Completed: 340 Avg. Call Rate: 0.615/sec Errors Detected: 0 Error Rate: 0.000% Registered: 0

The recent calls view is designed to provide an in-depth view of each recent VoIP call and its status. Each call is represented by an entry in the topmost report; the list is updated as the calls are removed from the active calls view.

This view contains the following columns:

Call Status: The current status of the call. These may be things such as connecting, ringing, connected, error, etc.

Protocol: The values for this field are SIP or H.323.

Started: This is the time (local time) that the call was started.

Duration: The length of time the call is (or was) active.

Terminator: Which side of the call (Source or Destination) terminated the call.

Source Address: The address of the call initiator (caller).

Source ID/E.164: The SIP user ID or H.323 E.164 alias of the caller.

Source Name/H.323 ID: The SIP display name or H.323 ID of the caller.

Destination Address: The address of the call receiver (party called).

Destination ID/E.164: The SIP user ID or H.323 E.164 alias of the party called.

Destination Name/H.323 ID: The SIP display name or H.323 ID of the party called.

Call ID: The SIP or H.323 call ID associated with this call.

Registered With: The gatekeeper's IP address for H.323 calls, or the Proxy's IP address for SIP calls.

Conference ID: The conference ID (H.323 calls only).

Each individual call has the following sub-views.

- Call Summary
- Call Flow (ladder diagram)
- Call Trace
- Call Metrics
- Audio Details
- Audio QoS
- Video Details
- Data Details
- RTCP Summary
- RTCP XR Summary
- DTMF Summary

Call Summary

Call Summary Call Flow Call Trace Call Metrics Audio Details Audio QoS Video Details Data Details RTPCP Summary RTPCP/RTT Summary DTMF Summary			
Signaling		Value	
Src Address	120.249.49.11		
Src E.164	1004		
Src H.323 ID			
Dest Address	120.249.49.21		
Dest E.164	2004		
Dest H.323 ID			
Start Time	09:20:12		
Stop Time	09:21:12		
Duration	00:01:00		
Call Terminator	Source		
Gatekeeper			
Call Score	A+	99.533	
Signaling Score	A+	99.517	
Media Score	A+	99.537	
Src Audio Score	A+	99.736	
Dest Audio Score	A+	99.532	
Src Vid Score	A+	99.508	
Dest Vid Score	A+	99.276	
Record Filename			
Capture Filename			
Audio		Value	
Src Audio Channel	120.249.49.11:25176		
Src Media Type	G.729		
Src Packet Count	1,000		
Src Average Jitter (ms)	0.596		
Src Average Packet Interval (ms)	60.061		
Src Average Bandwidth (kb/s)	8.036		
Src Packets Lost	0		
Src TOS/DSCP Flag	Default (000000)		
Src Listening R-Factor	63		
Src Listening MOS Score	3.95		
Src Conversational R-Factor	62		
Src Conversational MOS Score	3.91		
Stream Quality Index (SQI)	A+ (100.00)		
Dest Audio Channel	120.249.49.21:25176		
Dest Media Type	G.729		
Dest Packet Count	1,000		
Dest Average Jitter (ms)	1.815		
Dest Average Packet Interval (ms)	60.058		
Dest Average Bandwidth (kb/s)	8.036		
Dest Packets Lost	0		
Dest TOS/DSCP Flag	Default (000000)		
Dest Listening R-Factor	63		
Dest Listening MOS Score	3.95		
Dest Conversational R-Factor	62		
Dest Conversational MOS Score	3.91		
Stream Quality Index (SQI)	A+ (100.00)		
Video		Value	
Src Video Channel	120.249.49.11:25178		
Src Media Type	H.263		
Src Packet Count	909		
Src Average Jitter (ms)	0.575		
Src Average Packet Interval (ms)	66.074		
Src Average Bandwidth (kb/s)	8.857		
Src Packets Lost	0		
Src TOS/DSCP Flag	Default (000000)		
Dest Video Channel	120.249.49.21:25178		
Dest Media Type	H.263		
Dest Packet Count	909		
Dest Average Jitter (ms)	0.879		
Dest Average Packet Interval (ms)	66.078		
Dest Average Bandwidth (kb/s)	8.856		
Dest Packets Lost	0		
Dest TOS/DSCP Flag	Default (000000)		

This sub-view provides a summary of the call elements including source and destination addresses for signaling and media. There are three panes on this sub-view.

Signaling Pane

Source Address: The IP address of the calling endpoint.

Source ID/Source E.164: The source ID (SIP) or E.164 (H.323) of the calling endpoint.

Source Name/Source H.323 ID: The source name (SIP) or H.323 ID (H.323) of the calling endpoint.

Destination Address: The IP address of the called endpoint.

Destination ID/E.164: The source ID (SIP) or E.164 (H.323) of the called endpoint.

Destination Name/H.323 ID: The source name (SIP) or H.323 ID (H.323) of the called endpoint.

Start Time: The time the first packet was seen on the network.

Stop Time: The time the last packet was seen on the network.

Duration: The difference between the start and stop time.

Call Terminator: The endpoint that terminated the call.

Proxy/Gatekeeper: The address of the proxy (SIP) or gatekeeper (H.323) that participated in the call.

Call Score: Please see Appendix C for scoring details.

Signal Score: Please see Appendix C for scoring details.

Media Score: Please see Appendix C for scoring details.

Source Audio Score: Please see Appendix C for scoring details.

Destination Audio Score: Please see Appendix C for scoring details.

Source Video Score: Please see Appendix C for scoring details.

Destination Video Score: Please see Appendix C for scoring details.

Record Filename: The file name of the record file.

Capture Filename: The file name of the capture file.

Audio Pane

Source Audio Channel: The IP address and port of the calling endpoint.

Source Media Type: The type of codec being used to send the audio.

Source Packet Count: The number of packets sent on this channel.

Source Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Source Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

Source Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Source Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Source TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

Source Listening R Factor: The current Listening R factor for this media stream.

Source Listening MOS Score: The current Listening MOS Score for this media stream.

Source Conversational R Factor: The current Conversational R factor for this media stream.

Source Conversational MOS Score: The current Conversational MOS Score for this media stream.

Stream Quality Index: The ratio of the current R Factor and MOS score to there optimal values.

Destination Audio Channel: The IP address and port of the called endpoint.

Destination Media Type: The type of codec being used to send the audio.

Destination Packet Count: The number of packets sent on this channel.

Destination Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Destination Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

Destination Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Destination Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Destination TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

Destination Listening R Factor: The current Listening R factor for this media stream.

Destination Listening MOS Score: The current Listening MOS Score for this media stream.

Destination Conversational R Factor: The current Conversational R factor for this media stream.

Destination Conversational MOS Score: The current Conversational MOS Score for this media stream.

Stream Quality Index: The ratio of the current R Factor and MOS score to there optimal values.

Video Pane

Source Video Channel: The IP address and port of the calling endpoint.

Source Media Type: The type of codec being used to send the Video.

Source Packet Count: The number of packets sent on this channel.

Source Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Source Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

Source Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Source Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Source TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

Destination Video Channel: The IP address and port of the called endpoint.

Destination Media Type: The type of codec being used to send the Video.

Destination Packet Count: The number of packets sent on this channel.

Destination Average Jitter (ms): The average jitter value (as calculated from RFC 3550) for this channel.

Destination Average Packet Interval (ms): The average inter-arrival time of packets on this channel.

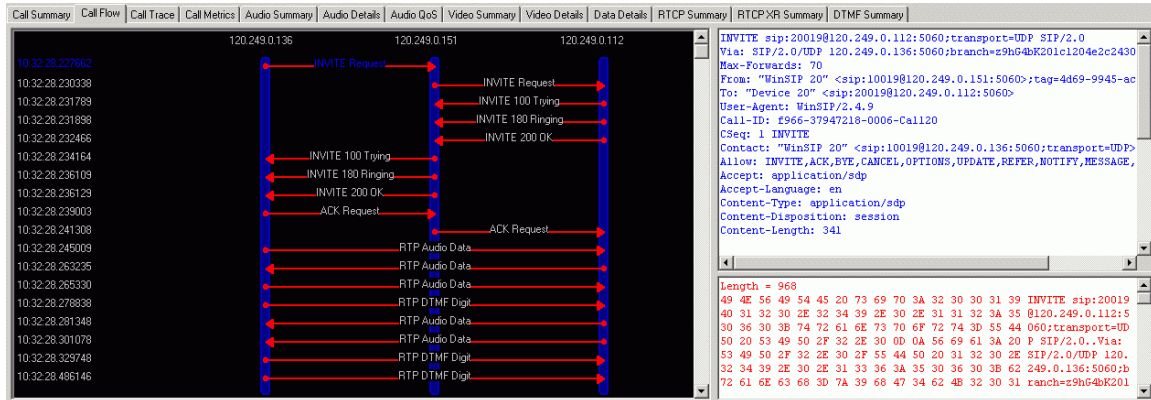
Destination Average Bandwidth (kb/s): The average bandwidth, in kilobits per second, calculated for this channel.

Destination Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received.

Destination TOS/DSCP Flag: The value of the Type of Service (TOS) / Differentiated Services Code Point (DSCP) flag in the IP header field.

WinEyeQ User's Guide

Call Flow



This sub-view provides a time-stamped ladder diagram view of the call flow (signaling, media and media quality packets). Each “rung” in the ladder may be highlighted to display the decoded packet in both ASCII and hexadecimal representations.

Call Trace

Timestamp	Source IP	Port	Protocol	Method	Type	Code	Text	Dest IP	Port
10:38:34.448453	120.249.0.136	20008	SIP	INVITE	Request			120.249.0.151	5060
10:38:34.451944	120.249.0.151	5060	SIP	INVITE	Request			120.249.0.112	5060
10:38:34.452863	120.249.0.112	20010	SIP	INVITE	Response	100	Trying	120.249.0.151	5060
10:38:34.452867	120.249.0.112	20010	SIP	INVITE	Response	180	Ringing	120.249.0.151	5060
10:38:34.452916	120.249.0.112	20010	SIP	INVITE	Response	200	OK	120.249.0.151	5060
10:38:34.452916	120.249.0.151	5060	SIP	INVITE	Response	100	Trying	120.249.0.136	5060
10:38:34.455682	120.249.0.151	5060	SIP	INVITE	Response	180	Ringing	120.249.0.136	5060
10:38:34.456774	120.249.0.151	5060	SIP	INVITE	Response	200	OK	120.249.0.136	5060
10:38:34.457845	120.249.0.136	20008	SIP	ACK	Request			120.249.0.151	5060
10:38:34.459666	120.249.0.151	5060	SIP	ACK	Request			120.249.0.112	5060
10:38:34.460622	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.465518	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.479560	120.249.0.112	40068	RTP					120.249.0.136	40068
10:38:34.484734	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.499419	120.249.0.112	40068	RTP					120.249.0.136	40068
10:38:34.518951	120.249.0.112	40068	RTP					120.249.0.136	40068
10:38:34.577471	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.733939	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.785406	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.894987	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:34.945757	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.100960	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.151880	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.304035	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.354632	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.514801	120.249.0.136	40068	RTP					120.249.0.112	40068
10:38:35.565764	120.249.0.136	40068	DTM					120.249.0.112	40068

Packet Details (Right Pane):

```
INVITE sip:20017@120.249.0.112:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 120.249.0.136:5060;branch=z9hG4bK20003024e28248a
Max-Forwards: 70
From: "WinSIP 18" <sip:10017@120.249.0.151:5060>;tag=415a-eadc-53
To: "Device 18" <sip:20017@120.249.0.112:5060>
User-Agent: WinSIP/2.4.9
Call-ID: 1749-38313437-000c-Call118
CSeq: 1 INVITE
Contact: "WinSIP 18" <sip:10017@120.249.0.136:5060;transport=UDP>
Allow: INVITE,ACK,BYE,CANCEL,OPTIONS,UPDATE,REFER,NOTIFY,MESSAGE,
Accept: application/sdp
Accept-Language: en
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 341

Length = 968
49 4E 56 49 54 45 20 73 69 70 3A 32 30 30 31 37 INVITE sip:20017
40 31 32 30 2E 32 34 39 2E 30 2E 31 31 32 3A 35 8120.249.0.112:5
30 36 30 3B 74 72 61 6E 73 70 6F 72 74 3D 55 44 060;transport=UD
50 20 53 49 50 2F 32 2E 30 0D 0A 56 69 61 3A 20 P SIP/2.0. Via:
53 49 50 2F 32 2E 30 2F 55 44 50 20 31 32 30 2E SIP/2.0/UDP 120.
32 34 39 2E 30 2E 31 33 36 3A 35 30 36 30 3B 62 249.0.136:5060;b
72 61 6E 63 68 3D 7A 39 68 47 34 62 4B 32 30 30 ranch=z9hG4bK200
```

This sub-view provides a time stamped protocol specific display of the call flow (signaling, media and media quality packets). Each entry in the report may be highlighted to display the decoded packet in both ASCII and hexadecimal representations.

This sub-view provides a summary of the call elements including metric measurements for response times and signaling interval and packets and byte counts analyzed by each protocol layer. There are two panes on this sub-view.

Initial Response Time: The length of time it took for the first message that was sent by the calling endpoint to be acknowledged by the called endpoint or proxy/gatekeeper.

Ring Duration: The length of time the call was ringing.

Time to Connect: The length of time it took for the call to be connected.

Time Connected: The length of time from when the call was connected until the close sequence started.

Signal Latency: The length of time it took for the call to connect plus disconnect.

Source Audio Delay: The length of time from when the call was connected until the first source audio packet was sent.

Source Video Delay: The length of time from when the call was connected until the first source video packet was sent.

Destination Audio Delay: The length of time from when the call was connected until the first destination audio packet was sent.

Destination Video Delay: The length of time from when the call was connected until the first destination video packet was sent.

Protocol Pane

Packets and Bytes count, for the selected call, broken down by indicated protocol.

Audio Details

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTP Summary RTP>R Summary DTMF Summary						
Metric	Current	Low	High	Parameter	Source	Destination
Src Jitter (ms)	4.408	0.161	9.019	Address	120.249.1.13	120.249.1.12
occurred at:		00:00:01	00:00:04	Port	40052	40052
Dest Jitter (ms)	3.169	0.752	5.977	Media Type	G.711 Alaw	G.711 Alaw
occurred at:		00:00:01	00:00:04	SSRC	2870DACD	A6C12802
Avg Src Packet Interval (ms)	20.124	2.137	77.041	Audio/Package (ms)	20	20
occurred at:		00:00:04	00:00:02	Frames/Package	20	20
Avg Dest Packet Interval (ms)	20.106	0.949	39.103	Total Packets	237	237
occurred at:		00:00:04	00:00:04	Packets Lost	0	0
Avg Src Bandwidth (kb/s)	64.761	64.286	66.667	Early Packets	76	42
occurred at:		00:00:05	00:00:01	Late Packets	58	41
Avg Dest Bandwidth (kb/s)	64.815	64.214	66.667	DTMF Events		
occurred at:		00:00:05	00:00:01	Current Bandwidth (kb/s)	62.49	62.49
				Longest Packet Loss Burst	0	0
				Total Payload Bytes	37,920	37,920

This sub-view provides summary information including jitter and interval measurements for the audio streams. The high, low and current values for each stream as well as the stream type, the sender's IP address and port, the receiver's IP address and port, the number of packets lost and the DTMF sequences if present within the stream (RFC 2833 section 3 Named Telephony Events). This sub-view contains two panes.

Metrics Pane

Source Jitter (ms): The average, low, and high jitter measurements calculated for the source audio stream.

Occurred at: The time, relative to the start of the call, that the high and low source jitter values were calculated.

Destination Jitter (ms): The average, low, and high jitter measurements calculated for the destination audio stream.

Occurred at: The time, relative to the start of the call, that the high and low destination jitter values were calculated.

Average Source Packet Interval (ms): The average, low, and high inter-arrival time of packets on the source audio stream.

Occurred at: The time, relative to the start of the call, that the high and low source packet interval values were calculated.

Average Destination Packet Interval (ms): The average, low, and high inter-arrival time of packets on the destination audio stream.

Occurred at: The time, relative to the start of the call, that the high and low destination packet interval values were calculated.

Average Source Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the source audio stream.

Occurred at: The time, relative to the start of the call, that the high and low source bandwidth values were calculated.

Average Destination Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the destination audio stream.

Occurred at: The time, relative to the start of the call, that the high and low destination bandwidth values were calculated.

Parameters Pane

Address: The IP addresses for the source and destination channels.

Port: The port numbers for the source and destination channels.

Media Type: The codec type for the source and destination channels.

SSRC: The synchronization source for the source and destination channels.

Audio/Packet (ms): The length of audio time contained in each packet for the source and destination channels.

Frames/Packet: The number of audio frames contained in each packet for the source and destination channels.

Total Packets: The number of packets counted for the source and destination channels.

Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received for the source and destination channels.

Early Packets: The number of packets considered early as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

Late Packets: The number of packets considered late as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

DTMF Events: The value of the DTMF digits (RFC 2833) for the source and destination channels.

Current Bandwidth (kb/s): The bandwidth, in kilobits per second, calculated during the last second for the source and destination channels.

Longest Packet Loss Burst: The count of the longest sequence of lost packets for the source and destination channels.

Total Payload Bytes: The number of bytes in the payload portion of the packet for the source and destination channels.

Audio QoS

Call Summary Call Flow Call Trace Call Metrics Audio Details Audio QoS Video Details Data Details RTP Summary RTP/RTT Summary DTMF Summary			
Metric	Source	Destination	
Optimal Listening R Factor	93	93	
Actual Listening R Factor	90	93	
Conversational R Factor	89	92	
Optimal Listening MOS Score	4.20	4.20	
Actual Listening MOS Score	4.14	4.20	
Conversational MOS Score	4.11	4.18	
P.862 Raw MOS Score	4.07	4.45	
Stream Quality Index (SQI)	A (97.67)	A+ (100.00)	
Packet Loss Rate (%)	0.00	0.00	
Burst Packet Loss Rate (%)	0.04	0.00	
Good Packets	2,992	2,999	
Lost Packets	0	0	
Discarded Packets	9	2	
Media Type	G.711 Alaw	G.711 Alaw	
Jitter Buffer Min Delay (ms)	10	10	
Jitter Buffer Max Delay (ms)	75	40	
Jitter Buffer Avg Delay (ms)	40	34	
Jitter Buffer Current Delay (ms)	10	10	
Jitter Buffer Mode	Adaptive	Adaptive	

This sub-view provides the QoS metrics for each stream. The R-factor/MOS scoring feature is a non-intrusive measurement technique available for the WinEyeQ call monitor. WinEyeQ passively measures the characteristics of live VoIP calls and reports quality scores in real-time. The algorithm used to obtain the R-Factor/MOS quality scores accurately models the way that time-varying impairments, most notably burst packet loss and possible jitter buffer discards, affect perceived speech quality. This sub-view contains a single pane.

Metrics Pane

Optimal Listening R Factor: The highest score that is attainable for this codec.

Actual Listening R Factor: The current value of the Listening R Factor for the source and destination audio streams.

Conversational R Factor: The current value of the Conversational R Factor for the source and destination audio streams.

Optimal Listening MOS Score: The highest score that is attainable for this codec.

Actual Listening MOS Score: The current value of the Listening MOS Score for the source and destination audio streams.

Conversational MOS Score: The current value of the Conversational MOS Score for the source and destination audio streams.

P.862 Raw MOS Score: The current value of the P.862 Raw MOS Score for the source and destination audio streams.

Stream Quality Index: The ratio of the current R Factor and MOS score to there optimal values.

Packet Loss Rate (%): The total number packets that were lost divided by the total number of packets that were received.

Burst Packet Loss Rate (%): The packet loss rate encountered for burst conditions for the source and destination audio streams.

Good Packets: The number of packets received for the source and destination audio streams.

Lost Packets: The number of network lost packets for the source and destination audio streams.

Discarded Packets: The number of discarded packets due to excessive delay or extremely early arrival detected for the source and destination audio streams.

Media Type: The codec type for the source and destination audio streams.

Jitter Buffer Minimum Delay (ms): The minimum jitter buffer emulator delay in milliseconds occurring during a call for the source and destination audio streams.

Jitter Buffer Maximum Delay (ms): The maximum jitter buffer emulator delay in milliseconds occurring during a call for the source and destination audio streams.

Jitter Buffer Average Delay (ms): The average jitter buffer emulator delay in milliseconds occurring during a call for the source and destination audio streams.

Jitter Buffer Current Delay (ms): The current jitter buffer emulator delay in milliseconds for the source and destination audio streams.

Jitter Buffer Mode: The type of jitter buffer (adaptive or fixer) being used for the source and destination audio streams. This is configured on the Edit Menu | Settings | QoS Tab.

The quality scores for MOS range from 0 to 4.5 and the R factor measurements range from 0 to 105 depending on codec type. The guidelines for interpreting the R-factor and MOS scores are shown in the table below for the G.711 codec:

Desirability Scale	R-factor Range	MOS Range
Desirable	94 - 80	4.4 - 4.0
Acceptable	80 - 70	4.0 - 3.6
Reach Connection	70 - 50	3.6 - 2.6
Not recommended	50 - 0	2.6 - 0

Video Details

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTP Summary RTP XR Summary DTMF Summary						
Metric	Current	Low	High	Parameter	Source	Destination
Src Jitter (ms)	0.768	0.054	2.804	Address	120.249.1.13	120.249.1.11
occurred at:		00:00:01	00:00:30	Port	50354	50554
Dest Jitter (ms)	0.923	0.028	2.028	Media Type	H.263	H.263
occurred at:		00:00:01	00:00:32	SSRC	D63BCD8	32736D1C
Avg Src Packet Interval (ms)	66.484	51.965	79.324	Total Packets	904	909
occurred at:		00:00:30	00:00:30	Packets Lost	0	0
Avg Dest Packet Interval (ms)	66.077	56.873	75.121	Early Packets	8	17
occurred at:		00:00:30	00:00:30	Late Packets	8	12
Avg Src Bandwidth (kb/s)	121.864	121.465	125.648	Pictures	904	909
occurred at:		00:00:56	00:00:01	Picture Rate	15.092	15.184
Avg Dest Bandwidth (kb/s)	8.438	8.438	11.939	Current Bandwidth (kb/s)	0.00	0.00
occurred at:		00:00:53	00:00:01	Longest Packet Loss Burst	0	0
				Total Payload Bytes	910,687	64,113

This sub-view provides summary information including jitter and interval measurements for the video streams. The measurements include high, low, and current values for each stream. As well as the stream type, the sender's IP address and port number, the receiver's IP address and port number, the packets lost, the number of pictures detected, and the picture rate. This sub-view contains two panes.

Metrics Pane

Source Jitter (ms): The average, low, and high jitter measurements calculated for the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source jitter values were calculated.

Destination Jitter (ms): The average, low, and high jitter measurements calculated for the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination jitter values were calculated.

Average Source Packet Interval (ms): The average, low, and high inter-arrival time of packets on the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source packet interval values were calculated.

Average Destination Packet Interval (ms): The average, low, and high inter-arrival time of packets on the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination packet interval values were calculated.

Average Source Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source bandwidth values were calculated.

Average Destination Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination bandwidth values were calculated.

Parameters Pane

Address: The IP addresses for the source and destination channels.

Port: The port numbers for the source and destination channels.

Media Type: The codec type for the source and destination channels.

SSRC: The synchronization source for the source and destination channels.

Total Packets: The number of packets counted for the source and destination channels.

Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received for the source and destination channels.

Early Packets: The number of packets considered early as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

Late Packets: The number of packets considered late as configured on the Edit Menu | Settings | Advanced Tab for the source and destination channels.

Pictures: The number of picture start codes counted for the source and destination channels.

Picture Rate: The number of pictures per second calculated for the source and destination channels.

Current Bandwidth (kb/s): The bandwidth, in kilobits per second, calculated during the last second for the source and destination channels.

Longest Packet Loss Burst: The count of the longest sequence of lost packets for the source and destination channels

Total Payload Bytes: The number of bytes in the payload portion of the packet for the source and destination channels.

Data Details

[illegible]

This sub-view provides summary information including interval measurements for the data streams, the high, low and current values for each stream as well as the stream type, the sender's IP address and port, the receiver's IP address and port, and the number of packets lost. There are two panes on this sub-view.

Metrics Pane

Average Source Packet Interval (ms): The average, low, and high inter-arrival time of packets on the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source packet interval values were calculated.

Average Destination Packet Interval (ms): The average, low, and high inter-arrival time of packets on the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination packet interval values were calculated.

Average Source Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the source video stream.

Occurred at: The time, relative to the start of the call, that the high and low source bandwidth values were calculated.

Average Destination Bandwidth: The average, low, and high bandwidth, in kilobits per second, calculated for the destination video stream.

Occurred at: The time, relative to the start of the call, that the high and low destination bandwidth values were calculated.

WinEyeQ User's Guide

Parameters Pane

Address: The IP addresses for the source and destination channels.

Port: The port numbers for the source and destination channels.

Media Type: The codec type for the source and destination channels.

SSRC: The synchronization source for the source and destination channels.

Total Packets: The number of packets counted for the source and destination channels.

Packets Lost: The calculated number of packets lost by subtracting the number of packets expected (using the sequence numbers) minus the number actually received for the source and destination channels.

Current Bandwidth (kb/s): The bandwidth, in kilobits per second, calculated during the last second for the source and destination channels.

Longest Packet Loss Burst: The count of the longest sequence of lost packets for the source and destination channels

Total Payload Bytes: The number of bytes in the payload portion of the packet for the source and destination channels.

RTCP Summary

Call Summary Call Flow Call Trace Call Metrics Audio Summary Audio Details Audio QoS Video Summary Video Details Data Details RTCP Summary RTCP/VR Summary DTMF Summary			
Source RTCP Channel Summary			
Metric	Audio Channel	Video Channel	
Sender Address	120.249.49.11:29889	120.249.49.11:29891	
Receiver Address	120.249.0.150:26593	120.249.0.150:26595	
Sender Reports	1	1	
Receiver Reports	1	1	
SDS Reports	1	1	
Bye Reports	0	0	
Application Reports	0	0	
Senders RTP Packet Count	6	5	
Senders RTP Byte Count	2,880	1,401	
Reported Jitter (ms)	0.00	0.00	
Delay Since Last SR (sec)	0.00	0.00	
Reported Packets Lost	0	0	
Highest Sequence Number	0	0	
Fraction Lost (%)	0.00	0.00	
Canonical Name	Win323 Initiate Call 1	Win323 Initiate Call 1	
Name	Audio	Video	
E-Mail Address	support@touchstone-inc.com	support@touchstone-inc.com	
Phone Number	+215-672-6950	+215-672-6950	
Location	228 North York Road, Suite D, Hatboro, PA 19040 ...	228 North York Road, Suite D, Hatboro, PA 19040 ...	
Tool	Voice And Video Over IP Test Tool	Voice And Video Over IP Test Tool	
Note	Version 1.5.0	Version 1.5.0	
Private	www.touchstone-inc.com	www.touchstone-inc.com	
Destination RTCP Channel Summary			
Metric	Audio Channel	Video Channel	
Sender Address	120.249.0.150:26593	120.249.0.150:26595	
Receiver Address	120.249.49.11:29889	120.249.49.11:29891	
Sender Reports	1	1	
Receiver Reports	1	1	
SDS Reports	1	1	
Bye Reports	0	0	
Application Reports	0	0	
Senders RTP Packet Count	37	123	
Senders RTP Byte Count	17,760	125,251	
Reported Jitter (ms)	0.00	0.00	
Delay Since Last SR (sec)	0.00	0.00	
Reported Packets Lost	0	0	
Highest Sequence Number	0	0	
Fraction Lost (%)	0.00	0.00	
Canonical Name	Win323 Terminate Call 31624	Win323 Terminate Call 31624	
Name	Audio	Video	
E-Mail Address	support@touchstone-inc.com	support@touchstone-inc.com	
Phone Number	+215-672-6950	+215-672-6950	
Location	228 North York Road, Suite D, Hatboro, PA 19040 ...	228 North York Road, Suite D, Hatboro, PA 19040 ...	
Tool	Voice And Video Over IP Test Tool	Voice And Video Over IP Test Tool	
Note	Version 1.5.3	Version 1.5.3	
Private	www.touchstone-inc.com	www.touchstone-inc.com	

This sub-view provides summary information that has been gathered from the RTCP packets that WinEyeQ has analyzed for the audio and video streams that have been sent by both endpoints of the call. This sub-view has two panes that are identical except for source and destination.

RTCP Channel Summary Pane

Sender Address: The IP address and port number of the sending RTCP channel.

Receiver Address: The IP address and port number of the receiving RTCP channel.

Sender Reports: The number of RTCP Sender Reports sent.

Receiver Reports: The number of RTCP Receiver Reports sent.

SDES: The number of RTCP SDES Reports sent.

Bye Reports: The number of RTCP Bye Reports sent.

Application Reports: The number of RTCP Application Reports sent.

Senders Packet Count: The total number of RTP data packets transmitted by the sender since starting transmission.

Senders Byte Count: The total number of payload octets transmitted in RTP data packets by the sender since starting transmission.

Reported Jitter (ms): The jitter measurement calculated on the stream being received from the other endpoint.

Delay Since Last SR (sec): The delay, expressed in units of 1/65536 seconds, between receiving the last SR packet from the remote endpoint and sending this reception report block.

Reported Packets Lost: The total number of RTP data packets from the remote endpoint that have been lost since the beginning of reception.

Highest Sequence Number: The low 16 bits contain the highest sequence number received in an RTP data packet from the remote endpoint, and the most significant 16 bits extend that sequence number with the corresponding count of sequence number cycles.

Fraction Lost (%): The fraction of RTP data packets from the remote endpoint lost since the previous SR or RR packet was sent, expressed as a fixed point number with the binary point at the left edge of the field.

Canonical Name: A unique end-point identifier.

Name: The real name used to describe the source.

E-Mail Address: The email address is formatted according to RFC 2822.

Phone Number: The phone number (should be formatted with the plus sign replacing the international access code).

Location: The geographic user location.

Tool: The application or tool name.

Note: This is intended for transient messages describing the current state of the source.

Private: This item is used to define experimental or application-specific extensions.

RTCP XR Summary

Cal Summary Cal Flow Cal Trace Cal Metrics Audio Summary Audio Di Audio QoS Video Summary Video Details Data Tables RTPCVR Summary RTPCVR Summary DTMF Summary											
Source RTPCVR Channel Summary:				Destination RTPCVR Channel Summary:							
Metric		Audio Channel		Video Channel		Metric		Audio Channel		Video Channel	
Sender Address		120.249.49.10:25001		120.249.49.10:25003		Sender Address		120.249.0.150:25041		120.249.49.10:25043	
Receiver Address		120.249.0.150:25004		120.249.0.150:25043		Receiver Address		120.249.49.10:25001		120.249.49.10:25003	
Extended Reports		22		22		Extended Reports		22		22	
Loss Rate (%)		14.45		14.45		Loss Rate (%)		14.45		14.45	
Discard Rate (%)		0.00		0.00		Discard Rate (%)		0.00		0.00	
Average Burst Density (%)		19.53		19.53		Average Burst Density (%)		19.53		19.53	
Average Gap Density (%)		0.00		0.00		Average Gap Density (%)		0.00		0.00	
Average Burst Duration (ms)		1,444		1,444		Average Burst Duration (ms)		1,444		1,444	
Average Gap Duration (ms)		445		445		Average Gap Duration (ms)		445		445	
Round Trip Delay (ms)		16		16		Round Trip Delay (ms)		16		16	
End System Delay (ms)		60		60		End System Delay (ms)		60		60	
Signal Level (db)		-105		-105		Signal Level (db)		-105		-105	
Noise Level (db)		-67		-67		Noise Level (db)		-67		-67	
Residual Echo Return Loss (db)		55		55		Residual Echo Return Loss (db)		55		55	
Gap Threshold		16		16		Gap Threshold		16		16	
R Factor		90		90		R Factor		90		90	
External R Factor		Unavailable		Unavailable		External R Factor		Unavailable		Unavailable	
Listening MOS		4.0		4.0		Listening MOS		4.0		4.0	
Conversational MOS		4.0		4.0		Conversational MOS		4.0		4.0	
Packet Loss Concealment		Enhanced		Enhanced		Packet Loss Concealment		Enhanced		Enhanced	
Jitter Buffer Adaptive		Adaptive		Adaptive		Jitter Buffer Adaptive		Adaptive		Adaptive	
Jitter Buffer Rate		8		8		Jitter Buffer Rate		8		8	
Jitter Buffer Nominal Delay (ms)		40		40		Jitter Buffer Nominal Delay (ms)		40		40	
Jitter Buffer Max Delay (ms)		40		40		Jitter Buffer Max Delay (ms)		40		40	
Jitter Buffer Absolute Max Delay (ms)		80		80		Jitter Buffer Absolute Max Delay (ms)		80		80	

This sub-view provides summary information that has been gathered from the RTCP XR packets that WinEyeQ has analyzed for the audio and video streams that have been sent by both endpoints of the call. This sub-view has two panes that are identical.

RTCP XR Channel Summary

Sender Address: The IP address and port number of the sending RTCP channel.

Receiver Address: The IP address and port number of the receiving RTP channel.

Extended Reports: The number of RTCP Extended Reports sent.

Loss Rate (%): The fraction of packets lost since the beginning of the call.

Discard Rate (%): The fraction of packets discarded since the beginning of the call.

Average Burst Density (%): The fraction of packets within burst periods since the beginning of the call.

Average Gap Density (%): The fraction of packets within gap periods since the beginning of the call.

Average Burst Duration (ms): The mean duration, in milliseconds, of the burst periods since the beginning of the call.

Average Gap Duration (ms): The mean duration, in milliseconds, of the gap periods since the beginning of the call.

Round Trip Delay (ms): The most recently calculated round-trip delay, in milliseconds.

End System Delay (ms): The most recently estimated end system delay, in milliseconds.

Signal Level (db): The relative speech signal level expressed as the ratio of the signal level to a 0 dBm0 reference.

Noise Level (db): The relative silence period noise level expressed as the ratio of the background noise level to a 0 dBm0 reference.

Residual Echo Return Loss (db): The residual echo return loss as the sum of the measured echo return loss (ERL) and the echo return loss enhanced (ERLE) of the echo canceller, expressed in dB.

Gap Threshold: The gap threshold, in packets.

R Factor: The voice quality metric for the call channel as measured in the monitored network segment.

External R Factor: The voice quality metric for the call channel as measured in an external monitored network segment.

Listening MOS: The estimated mean opinion listening quality score for the call channel.

Conversational MOS: The estimated mean opinion conversational quality score for the call channel.

Packet Loss Concealment: The packet loss concealment capabilities.

Jitter Buffer Adaptive: Adaptive or non-adaptive.

Jitter Buffer Rate: This represents the implementation specific adjustment rate of a jitter buffer in adaptive mode.

Jitter Buffer Nominal Delay (ms): The current nominal jitter buffer delay, in milliseconds.

Jitter Buffer Max Delay (ms): The maximum jitter buffer delay, in milliseconds, recorded for the call.

Jitter Buffer Absolute Max Delay (ms): The absolute maximum delay, in milliseconds, the jitter buffer can ever introduce to the call channel packet stream.

Recent Errors View

WinEyeQ Professional Edition w/QoS Version 1.7.0 on 120.249.49.10, 120.249.49.11 - Voice and Video Monitoring and Analysis

File Edit Capture Record View Help | Mode Analyze Network

Network Monitor Active Calls Recent Calls Recent Errors User Alerts User Alarms User Watches Endpoints Other Audio Other Video Registrations

Status	Protocol	Code	Description	Started	Duration	Source Address	Source ID/E.164	Source Name/H.323 ID	Destination Address	Destination ID/E.164	Destination Name/H.323 ID	Call ID
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10019	"WinSP 20"	120.249.49.20	20019	"Device 20"	900c-36759625-0001-Call20
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10017	"WinSP 18"	120.249.49.20	20017	"Device 18"	4f44-36759531-0001-Call18
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10015	"WinSP 16"	120.249.49.20	20015	"Device 16"	7632-36759421-0001-Call16
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10013	"WinSP 14"	120.249.49.20	20013	"Device 14"	466f-36759328-0001-Call14
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10011	"WinSP 12"	120.249.49.20	20011	"Device 12"	1e0d-36759218-0001-Call12
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10009	"WinSP 10"	120.249.49.20	20009	"Device 10"	25c8-36759125-0001-Call10
Error	SIP	608	Error Code 608	10:12:39		120.249.49.10	10007	"WinSP 8"	120.249.49.20	20007	"Device 8"	510c-36759015-0001-Call8
Error	SIP	608	Error Code 608	10:12:38		120.249.49.10	10005	"WinSP 6"	120.249.49.20	20005	"Device 6"	c6d6-36758921-0001-Call6
Error	SIP	608	Error Code 608	10:12:38		120.249.49.10	10003	"WinSP 4"	120.249.49.20	20003	"Device 4"	c7e7-36758812-0001-Call4
Error	SIP	608	Error Code 608	10:12:38		120.249.49.10	10001	"WinSP 2"	120.249.49.20	20001	"Device 2"	12d4-36758718-0001-Call2
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10018	"WinSP 19"	120.249.49.20	20018	"Device 19"	944a-36722703-0004-Call19
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10016	"WinSP 17"	120.249.49.20	20016	"Device 17"	e79b-36722593-0004-Call17
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10014	"WinSP 15"	120.249.49.20	20014	"Device 15"	ad6b-36722500-0004-Call15
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10012	"WinSP 13"	120.249.49.20	20012	"Device 13"	1091-36722390-0004-Call13
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10010	"WinSP 11"	120.249.49.20	20010	"Device 11"	1033-36722296-0004-Call11
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10008	"WinSP 9"	120.249.49.20	20008	"Device 9"	b26d-36722218-0004-Call9
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10007	"WinSP 8"	120.249.49.20	20007	"Device 8"	496f-36722156-0004-Call8
Error	SIP	606	Error Code 606	10:12:02		120.249.49.10	10004	"WinSP 5"	120.249.49.20	20004	"Device 5"	a5c0-36722046-0004-Call5
Error	SIP	606	Error Code 606	10:12:01		120.249.49.10	10002	"WinSP 3"	120.249.49.20	20002	"Device 3"	e991-36721921-0004-Call3
Error	SIP	606	Error Code 606	10:12:01		120.249.49.10	10001	"WinSP 2"	120.249.49.20	20001	"Device 2"	670c-36721820-0004-Call2
Error	SIP	600	Error Code 600	10:08:00		120.249.49.10	10017	"WinSP 18"	120.249.49.20	20017	"Device 18"	31f8-36480468-0008-Call18
Error	SIP	600	Error Code 600	10:08:00		120.249.49.10	10018	"WinSP 19"	120.249.49.20	20018	"Device 19"	36c3-36480406-0008-Call19
Error	SIP	600	Error Code 600	10:08:00		120.249.49.10	10013	"WinSP 14"	120.249.49.20	20013	"Device 14"	9b0a-36480265-0008-Call14
Error	SIP	600	Error Code 600	10:08:00		120.249.49.10	10012	"WinSP 13"	120.249.49.20	20012	"Device 13"	0237-36480156-0008-Call13
Error	SIP	600	Error Code 600	10:08:00		120.249.49.10	10010	"WinSP 11"	120.249.49.20	20010	"Device 11"	53b4-36480062-0008-Call11
Error	SIP	600	Error Code 600	10:08:00		120.249.49.10	10009	"WinSP 10"	120.249.49.20	20009	"Device 10"	ac91-36480031-0008-Call10

Call Summary | Call Flow | Call Trace | Call Metrics | Audio Details | Audio QoS | Video Details | Data Details | RTPC Summary | RTPCXR Summary | DTMF Summary

Signaling	Value	Audio	Value	Video	Value
Src Address	120.249.49.10	Src Audio Channel		Src Video Channel	
Src ID	10015	Src Media Type		Src Media Type	
Src Name	"WinSP 16"	Src Packet Count		Src Packet Count	
Dest Address	120.249.49.20	Src Average Jitter (ms)		Src Average Jitter (ms)	
Dest ID	20015	Src Average Packet Interval (ms)		Src Average Packet Interval (ms)	
Dest Name	"Device 16"	Src Average Bandwidth (kb/s)		Src Average Bandwidth (kb/s)	
Start Time	10:00:59	Src Packets Lost		Src Packets Lost	
Stop Time	10:00:59	Src TOSS/OSCP Flag		Src TOSS/OSCP Flag	
Duration		Src Listening R Factor			
Call Terminator	Unknown	Src Listening MOS Score			
Proxy		Src Conversational R Factor			
Call Score	A+ 99.643	Src Conversational MOS Score			
Signaling Score	A+ 99.643	Stream Quality Index (SQI)			
Media Score					
Src Aud Score		Dest Audio Channel		Dest Video Channel	
Dest Aud Score		Dest Media Type		Dest Media Type	
Src Vid Score		Dest Packet Count		Dest Packet Count	
Dest Vid Score		Dest Average Jitter (ms)		Dest Average Jitter (ms)	
Record Filename		Dest Average Packet Interval (ms)		Dest Average Packet Interval (ms)	
Capture Filename		Dest Average Bandwidth (kb/s)		Dest Average Bandwidth (kb/s)	
		Dest Packets Lost		Dest Packets Lost	
		Dest TOSS/OSCP Flag		Dest TOSS/OSCP Flag	
		Dest Listening R Factor			
		Dest Listening MOS Score			
		Dest Conversational R Factor			
		Dest Conversational MOS Score			
		Stream Quality Index (SQI)			

Ready: For Help, press F1 Session: 00:52:53 Active Calls: 31 Calls Completed: 2,090 Avg. Call Rate: 0.661/sec Errors Detected: 131 Error Rate: 6.260% Registered: 0

The recent errors view is designed to provide an in-depth view of each VoIP call for which an error was detected. Each call is represented by an entry, which is updated once every second, in the topmost report. This report contains the following columns:

Status: The current status of the call. These may be things such as connecting, ringing, connected, error, etc.

Protocol: The values for this field are SIP or H.323.

Code: The numeric code for this error (if applicable).

Description: A text description of this error.

Started: This is the time (local time) that the call was started.

Duration: The length of time the call is (or was) active.

Source Address: The address of the call initiator (caller).

Source ID/E.164: The SIP user ID or H.323 E.164 alias of the caller.

Source Name/H.323 ID: The SIP display name or H.323 ID of the caller.

Destination Address: The address of the call receiver (party called).

Destination ID/E.164: The SIP user ID or H.323 E.164 alias of the party called.

Destination Name/H.323 ID: The SIP display name or H.323 ID of the party called.

Call ID: The SIP or H.323 call ID associated with this call.

Registered With: The gatekeeper's IP address, for H.323 calls, or the Proxy's IP address, for SIP calls.

Conference ID: The conference ID (H.323 calls only).

Each individual call has the following sub-views.

- Call Summary
- Call Flow (ladder diagram)
- Call Trace
- Call Metrics
- Audio Details
- Audio QoS
- Video Details
- Data Details
- RTCP Summary
- RTCP XR Summary
- DTMF Summary

To display information about a particular call, select it (click the call line) in the call list. Whenever a call is selected, it will remain "locked" in the view for as long as you wish to view its details.

Note: For a detailed description of these sub-views please refer back to the 'Recent Calls View' section of the manual.

User Alerts View

WinEyeQ (tm) Professional Edition w/QoS Version 1.7.0 on 120.249.49.10, 120.249.49.11 - Voice and Video Monitoring and Analysis

File Edit Capture Record View Help

Mode: Analyze Network

Time	Notification	Threshold	Value	User ID	Source Address	Destination Address	Call/Conference ID
12:13:12.87775	Burst Packet Loss High Alert	1.00	2.00	10005	120.249.49.10:40020	120.249.0.150:47456	c41a-43991796-0547-Call6
12:13:15.24440	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.48	10024	120.249.49.10:40096	120.249.0.150:47532	11be-43994625-0547-Call25
12:13:18.89093	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.57	10015	120.249.49.10:40060	120.249.0.150:47592	6b16-43997750-0548-Call16
12:13:18.93452	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.57	10016	120.249.49.10:40064	120.249.0.150:47596	192a-43997796-0548-Call17
12:13:20.29452	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10010	120.249.49.10:40040	120.249.0.150:47608	998b-43998646-0548-Call11
12:13:20.29483	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10019	120.249.49.10:40076	120.249.0.150:47612	870e-43999000-0548-Call20
12:13:21.019626	Burst Packet Loss High Alert	1.00	2.00	10021	120.249.49.10:40084	120.249.0.150:47620	6a0b-43999187-0548-Call22
12:13:25.55036	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.39	20014	120.249.0.150:47728	120.249.49.10:40056	4b0a-44004670-0549-Call15
12:13:25.694989	Burst Packet Loss High Alert	1.00	2.00	10000	120.249.49.10:40000	120.249.0.150:47736	9550-44004687-054a-Call1
12:13:25.737097	Listening R Factor, (G.711 Alarm), Low Alert	83.70	83.00	20009	120.249.0.150:47676	120.249.49.10:40036	aab7-44002703-0549-Call10
12:13:25.737179	Listening R Factor, (G.711 Alarm), Low Alert	83.70	83.00	20011	120.249.0.150:47680	120.249.49.10:40044	3c3c-44002703-0549-Call12
12:13:25.737570	Listening R Factor, (G.711 Alarm), Low Alert	83.70	79.00	20017	120.249.0.150:47700	120.249.49.10:40068	f4f0-44004846-0549-Call18
12:13:25.783732	Listening R Factor, (G.711 Alarm), Low Alert	83.70	83.00	20012	120.249.0.150:47684	120.249.49.10:40048	0a67-44002703-0549-Call13
12:13:25.778950	Listening R Factor, (G.711 Alarm), Low Alert	83.70	83.00	20013	120.249.0.150:47688	120.249.49.10:40052	2e0d-44002796-0549-Call14
12:13:25.838473	Listening R Factor, (G.711 Alarm), Low Alert	83.70	82.00	20015	120.249.0.150:47692	120.249.49.10:40060	0e36-44002859-0549-Call16
12:13:25.878281	Listening R Factor, (G.711 Alarm), Low Alert	83.70	82.00	20016	120.249.0.150:47696	120.249.49.10:40064	f861-44002890-0549-Call17
12:13:25.897544	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.44	20018	120.249.0.150:47704	120.249.49.10:40072	355f-44004328-0549-Call19
12:13:31.460111	Burst Packet Loss High Alert	1.00	2.00	10018	120.249.49.10:40072	120.249.0.150:47804	b1b9-44009531-054a-Call19
12:13:32.870633	Bi-Directional Audio was not detected (from 120.249.0.150 to 120.249.49.10), High Alert	5.00	6.00	20001	120.249.49.10	120.249.0.150	5308-44007803-054a-Call2
12:13:32.913310	Bi-Directional Audio was not detected (from 120.249.0.150 to 120.249.49.10), High Alert	5.00	6.00	20007	120.249.49.10	120.249.0.150	f754-44007843-054a-Call8
12:13:39.433606	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.48	10001	120.249.49.10:40004	120.249.0.150:47964	1981-44010296-054c-Call2
12:13:39.473952	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10007	120.249.49.10:40028	120.249.0.150:47968	e19b-44010296-054c-Call8
12:13:41.953094	Burst Packet Loss High Alert	1.00	2.00	10000	120.249.49.10:40000	120.249.0.150:48036	1a01-44020966-054d-Call1
12:13:42.121396	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10001	120.249.49.10:40012	120.249.0.150:48048	1554-44021076-054d-Call4
12:13:44.040543	Burst Packet Loss High Alert	1.00	2.00	10001	120.249.49.10:40004	120.249.0.150:48064	e5d1-44023671-054d-Call2
12:13:46.613960	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10020	120.249.49.10:40080	120.249.0.150:48116	f0aa-44025953-054d-Call21
12:13:47.303023	Listening R Factor, (G.711 Alarm), Low Alert	83.70	81.00	20018	120.249.0.150:48104	120.249.49.10:40072	959e-44025281-054d-Call19
12:13:47.303080	Listening R Factor, (G.711 Alarm), Low Alert	83.70	81.00	20010	120.249.0.150:48108	120.249.49.10:40040	459f-44025296-054d-Call11
12:13:49.894610	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10005	120.249.49.10:40020	120.249.0.150:48156	796d-44027030-054e-Call6
12:13:50.073465	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10009	120.249.49.10:40036	120.249.0.150:48176	e72c-44028937-054e-Call10
12:13:50.474217	Burst Packet Loss High Alert	1.00	2.00	10017	120.249.49.10:40068	120.249.0.150:48200	311c-44030203-054e-Call18
12:13:50.984194	Burst Packet Loss High Alert	1.00	2.00	10011	120.249.49.10:40044	120.249.0.150:48180	a6c7-44028984-054e-Call12
12:13:53.333629	Listening MOS Score, (G.711 Alarm), Low Alert	3.78	3.53	10003	120.249.49.10:40012	120.249.0.150:48248	b827-44032000-054f-Call4

Alert Summary

Ready: For Help, press F1 Session: 02:01:14 Active Calls: 60 Calls Completed: 66,720 Avg. Call Rate: 9.449/sec Errors Detected: 0 Error Rate: 0.000% Registered: 0

This view provides an active list of the alerts that have occurred during the test session. The notification list for the events that triggered the alerts is displayed in tabular form. Each alert is represented by an entry in the topmost report. This report contains the following columns:

Time: This is the time the Alert was detected.

Notification: An explanation of the Alert.

Threshold: The threshold value set by the user.

Value: The value that triggered the Alert

User ID: The SIP user ID or H.323 alias of the caller.

Source Address: The address of the call initiator (caller).

Destination Address: The address of the call receiver (party called).

WinEyeQ User's Guide

Call/Conference ID: The SIP or H.323 call ID associated with this call.

Various alert thresholds are set by the user for audio/video jitter, interval, packet loss, and R-Factor/MOS score measurements

User Alarms View

Time	Notification	Threshold	Value	User ID	Source Address	Destination Address	Call/Conference ID
14:06:30.674882	Audio Jitter High Alarm Exceeded	2.00	2.07	10006	120.249.0.135:40024	120.249.0.111:49732	9ee#4077703-0013-Ca#7
14:06:32.457011	Video Jitter High Alarm Exceeded	2.00	2.06	20008	120.249.0.135:25674	120.249.0.112:28036	E64408CDB767C7F0D122B1B5C...
14:06:32.742861	Video Jitter High Alarm Exceeded	2.00	2.05	20002	120.249.0.135:25660	120.249.0.112:27992	810ED70C32E51AFPP2C054C9F3...
14:06:32.742941	Video Jitter High Alarm Exceeded	2.00	2.06	20004	120.249.0.135:25664	120.249.0.112:27996	82D47804B8F65F74314EC0208...
14:06:32.743021	Video Jitter High Alarm Exceeded	2.00	2.05	20006	120.249.0.135:25656	120.249.0.112:28000	096E3D1E2D37B6C399AE019972...
14:06:32.743091	Video Jitter High Alarm Exceeded	2.00	2.13	20005	120.249.0.135:25662	120.249.0.112:28004	F8651C027805FC2A1E602B1670E7...
14:06:32.743166	Video Jitter High Alarm Exceeded	2.00	2.20	20007	120.249.0.135:25666	120.249.0.112:28008	295253D7C058E654B1AK382C9...
14:06:32.743235	Video Jitter High Alarm Exceeded	2.00	2.29	20010	120.249.0.135:25670	120.249.0.112:28012	0648110DF88761184594AE831C16...
14:06:32.755143	Audio Jitter High Alarm Exceeded	2.00	2.02	10008	120.249.0.135:40032	120.249.0.111:49760	e701-50791937-0014-Ca#9
14:06:32.755213	Audio Jitter High Alarm Exceeded	2.00	2.02	10009	120.249.0.135:40036	120.249.0.111:49784	76ee-50791984-0014-Ca#10
14:06:32.791043	Audio Jitter High Alarm Exceeded	2.00	2.03	20005	120.249.0.135:25660	120.249.0.112:28002	F8651C027805FC2A1E602B1670E7...
14:06:32.791140	Audio Jitter High Alarm Exceeded	2.00	2.04	20007	120.249.0.135:25664	120.249.0.112:28006	295253D7C058E654B1AK382C9...
14:06:32.791212	Audio Jitter High Alarm Exceeded	2.00	2.04	20010	120.249.0.135:25660	120.249.0.112:28010	0648110DF88761184594AE831C16...
14:06:32.791294	Audio Jitter High Alarm Exceeded	2.00	2.05	20008	120.249.0.135:25672	120.249.0.112:28014	E64408CDB767C7F0D122B1B5C...
14:06:32.791366	Audio Jitter High Alarm Exceeded	2.00	2.08	20009	120.249.0.135:25676	120.249.0.112:28018	AFPP2C054C9F3D1E2D37B6C399...
14:06:32.801513	Video Jitter High Alarm Exceeded	2.00	2.27	20003	120.249.0.135:25644	120.249.0.112:27988	E0837A62FC064F360B72EC04833...
14:06:32.834840	Audio Jitter High Alarm Exceeded	2.00	2.10	10003	120.249.0.135:40012	120.249.0.111:49760	e131-50792515-0014-Ca#4
14:06:32.834911	Audio Jitter High Alarm Exceeded	2.00	2.09	10004	120.249.0.135:40016	120.249.0.111:49764	4984-50792562-0014-Ca#5
14:06:32.834970	Audio Jitter High Alarm Exceeded	2.00	2.08	10005	120.249.0.135:40020	120.249.0.111:49768	b253-50792687-0014-Ca#6
14:06:34.674872	Audio Jitter High Alarm Exceeded	2.00	2.05	10007	120.249.0.135:40028	120.249.0.111:49776	6383-50792796-0014-Ca#8
14:06:36.485380	Audio Jitter High Alarm Exceeded	2.00	2.33	10006	120.249.0.135:40024	120.249.0.111:49772	1715-50792765-0014-Ca#7
14:06:39.570407	Audio Jitter High Alarm Exceeded	2.00	2.02	20005	120.249.0.135:25700	120.249.0.112:28042	8E02B12549D0C039E16A4C40A97B...
14:06:39.570477	Audio Jitter High Alarm Exceeded	2.00	2.03	20010	120.249.0.135:25708	120.249.0.112:28050	FA143502F4E108774D3638414CE...
14:06:39.570569	Audio Jitter High Alarm Exceeded	2.00	2.03	20008	120.249.0.135:25712	120.249.0.112:28054	639171A7CB1504E3310A3A98A7...
14:06:39.570639	Audio Jitter High Alarm Exceeded	2.00	2.11	20009	120.249.0.135:25716	120.249.0.112:28058	AA608629F18879438B1D9C058D0...
14:06:39.884315	Audio Jitter High Alarm Exceeded	2.00	2.09	20007	120.249.0.135:25706	120.249.0.112:28046	239223751BFC3F1FC59616C99A48...
14:06:39.929615	Video Jitter High Alarm Exceeded	2.00	2.23	20006	120.249.0.135:25696	120.249.0.112:28040	8C28698020A3C6450E1142A2391...
14:06:39.929707	Video Jitter High Alarm Exceeded	2.00	2.41	20007	120.249.0.135:25704	120.249.0.112:28048	239223751BFC3F1FC59616C99A48...
14:06:39.929773	Video Jitter High Alarm Exceeded	2.00	4.49	20005	120.249.0.135:25702	120.249.0.112:28044	8C28612549D0C039E16A4C40A97B...
14:06:39.929843	Video Jitter High Alarm Exceeded	2.00	2.56	20010	120.249.0.135:25710	120.249.0.112:28052	FA143502F4E108774D3638414CE...
14:06:39.929920	Video Jitter High Alarm Exceeded	2.00	2.57	20008	120.249.0.135:25714	120.249.0.112:28056	639171A7CB1504E3310A3A98A7...
14:06:39.929960	Video Jitter High Alarm Exceeded	2.00	2.67	20009	120.249.0.135:25718	120.249.0.112:28060	AA608629F18879438B1D9C058D0...
14:06:39.930396	Audio Jitter High Alarm Exceeded	2.00	2.36	20002	120.249.0.135:25690	120.249.0.112:28030	E6F35508FC6E8A1B52E3AFAC0D0...

Alarm Summary

Ready: For Help, press F1

Session: 00:00:50

Active Calls: 40

Calls Completed: 50

Avg. Call Rate: 1.042/sec

Errors Detected: 0

Error Rate: 0.000%

Registered: 0

This view provides an active list of the alarms that have occurred during the test session. The notification list for the events that triggered the alarms is displayed in tabular form. Each alert is represented by an entry in the topmost report. This report contains the following columns:

Time: This is the time the Alarm was detected.

Notification: An explanation of the Alarm.

Threshold: The threshold value set by the user.

WinEyeQ User's Guide

Value: The value that triggered the Alarm

User ID: The SIP user ID or H.323 alias of the caller.

Source Address: The address of the call initiator (caller).

Destination Address: The address of the call receiver (party called).

Call/Conference ID: The SIP or H.323 call ID associated with this call.

Various alarm thresholds are set by the user for audio/video jitter, interval, packet loss, and R-Factor/MOS score measurements

WinEyeQ User's Guide

User Watches View

WinEyeQ (tm) Professional Edition w/Go5 Version 1.7.0 on 120.249.49.10, 120.249.49.11 - Voice and Video Monitoring and Analysis

File Edit Capture Record View Help

Mode Analyze Network

Network Monitor Active Calls Recent Calls Recent Errors User Alerts User Alarms User Watches Endpoints Other Audio Other Video Registrations

Status	Protocol	Found In	Watch Trigger	Started	Duration	Terminator	Source Address	Source ID/E.164	Source Name/H.323 ID	Destination Address	Destination ID/E.164	Destination Name/H.323 ID	Call ID
Setup	H.323	120.249.0.150	120.249.	12:21:40			120.249.49.11	1016		120.249.0.150	2016		00781344A708FC7560A67
Connected	H.323	120.249.0.150	120.249.	12:21:40			120.249.49.11	1018		120.249.0.150	2018		A91CD50690A210FCCECC0C
Connected	H.323	120.249.0.150	120.249.	12:21:40			120.249.49.11	1008		120.249.0.150	2008		957767A6A06A542C4244FC
Connected	H.323	120.249.0.150	120.249.	12:21:40			120.249.49.11	1022		120.249.0.150	2022		2D0CD6262E580893C43C4
Connected	H.323	120.249.0.150	120.249.	12:21:40			120.249.49.11	1009		120.249.0.150	2009		10F660B45A5A5778D46A80
Connected	H.323	120.249.0.150	120.249.	12:21:39			120.249.49.11	1019		120.249.0.150	2019		B1000328A23E3F7051608C
Connected	H.323	120.249.0.150	120.249.	12:21:39			120.249.49.11	1004		120.249.0.150	2004		3A1308F23A9AF3A7059994
Connected	SIP	sp:1001@...	120.249.	12:21:39			120.249.49.10	10010	"WinSP 11"	120.249.0.150	20010	"Device 11"	34B5-44493265-05a8-Call11
Connected	SIP	sp:1001@...	120.249.	12:21:39			120.249.49.10	10018	"WinSP 19"	120.249.0.150	20018	"Device 19"	769a-44499218-05a8-Call19
Completed	SIP	sp:1001@...	120.249.	12:21:38			120.249.49.10	10017	"WinSP 18"	120.249.0.150	20017	"Device 18"	8461-44498899-05a8-Call18
Released	H.323	120.249.0.150	120.249.	12:21:38			120.249.49.11	1014		120.249.0.150	2014		9596A2C01786CC6195003
Released	H.323	120.249.0.150	120.249.	12:21:38			120.249.49.11	1007		120.249.0.150	2007		6AB401E8C2915C4F9F8A638
Released	H.323	120.249.0.150	120.249.	12:21:38			120.249.49.11	1024		120.249.0.150	2024		9AFD07AB87A776FC31C4969
Completed	SIP	sp:20016@...	120.249.	12:21:38			120.249.49.10	10016	"WinSP 17"	120.249.0.150	20016	"Device 17"	6d7c-44498171-05a8-Call17
Completed	SIP	sp:1001@...	120.249.	12:21:38			120.249.49.10	10015	"WinSP 16"	120.249.0.150	20015	"Device 16"	8751-44498125-05a8-Call16
Released	H.323	120.249.0.150	120.249.	12:21:38			120.249.49.11	1011		120.249.0.150	2011		1FED458D6A6AC47E059421
Completed	SIP	sp:20013@...	120.249.	12:21:38			120.249.49.10	10013	"WinSP 14"	120.249.0.150	20013	"Device 14"	b1cc-44498031-05a8-Call14
Completed	SIP	sp:1001@...	120.249.	12:21:38			120.249.49.10	10012	"WinSP 13"	120.249.0.150	20012	"Device 13"	318b-44497904-05a8-Call13
Completed	SIP	sp:1001@...	120.249.	12:21:37			120.249.49.10	10011	"WinSP 12"	120.249.0.150	20011	"Device 12"	9187-44497937-05a8-Call12
Completed	SIP	sp:10009@...	120.249.	12:21:37			120.249.49.10	10009	"WinSP 10"	120.249.0.150	20009	"Device 10"	f0cb-44497975-05a8-Call10
Completed	SIP	sp:120.249.0.150	120.249.	12:21:37			120.249.49.10	10008	"WinSP 9"	120.249.0.150	20008	"Device 9"	0236-44497943-05a8-Call9
Completed	SIP	sp:20007@...	120.249.	12:21:37			120.249.49.10	10007	"WinSP 8"	120.249.0.150	20007	"Device 8"	cd21-44497796-05a8-Call8
Completed	SIP	sp:20001@...	120.249.	12:21:37			120.249.49.10	10001	"WinSP 2"	120.249.0.150	20001	"Device 2"	e284-44497734-05a8-Call2
Completed	SIP	sp:10006@...	120.249.	12:21:37			120.249.49.10	10006	"WinSP 7"	120.249.0.150	20006	"Device 7"	3364-44497607-05a8-Call7
Completed	SIP	sp:120.249.0.150	120.249.	12:21:37			120.249.49.10	10005	"WinSP 6"	120.249.0.150	20005	"Device 6"	2885-44497625-05a8-Call6
Completed	SIP	sp:20004@...	120.249.	12:21:37			120.249.49.10	10004	"WinSP 5"	120.249.0.150	20004	"Device 5"	6b30-44497609-05a8-Call5

Call Summary | Call Flow | Call Trace | Call Metrics | Audio Summary | Audio Details | Audio QoS | Video Summary | Video Details | Data Details | RTP/RTCP Summary | RTP/RTCP Summary | DTMF Summary

Signaling	Value	Audio	Value	Video	Value
Src Address	120.249.49.11	Src Audio Channel	120.249.49.11:27788	Src Video Channel	120.249.49.11:27790
Src E.164	1018	Src Media Type	G.711 Alaw 64k	Src Media Type	H.263
Src H.323 ID		Src Packet Count	84	Src Packet Count	76
Dest Address	120.249.0.150	Src Average Jitter (ms)	0.538	Src Average Jitter (ms)	0.429
Dest E.164	2018	Src Average Packet Interval (ms)	60.993	Src Average Packet Interval (ms)	67.313
Dest H.323 ID		Src Average Bandwidth (kb/s)	65.657	Src Average Bandwidth (kb/s)	10.722
Start Time	12:20:06	Src Packets Lost	0	Src Packets Lost	0
Stop Time	12:20:11	Src TOS/DSCP Flag	Default (000000)	Src TOS/DSCP Flag	Default (000000)
Duration	00:00:05	Src Listening R Factor	93		
Call Terminator	Source	Src Listening MOS Score	4.20		
Gatekeeper		Optimal Listening R Factor(MOS Score)	93% (20)		
Call Score	A+ 98.498	Stream Quality Index (SQI)	A+ (100.00)		
Signaling Score	A+ 98.633	Dest Audio Channel	120.249.0.150:29488	Dest Video Channel	120.249.0.150:29490
Media Score	A+ 98.614	Dest Media Type	G.711 Alaw 64k	Dest Media Type	H.263
Src Aud Score	A+ 99.548	Dest Packet Count	84	Dest Packet Count	273
Dest Aud Score	A+ 99.142	Dest Average Jitter (ms)	3.076	Dest Average Jitter (ms)	2.834
Src Vid Score	A+ 98.700	Dest Average Packet Interval (ms)	60.890	Dest Average Packet Interval (ms)	67.106
Dest Vid Score	A 96.336	Dest Average Bandwidth (kb/s)	45.580	Dest Average Bandwidth (kb/s)	515.499
Record Filename		Dest Packets Lost	0	Dest Packets Lost	0
Capture Filename		Dest TOS/DSCP Flag	Default (000000)	Dest TOS/DSCP Flag	Default (000000)
		Dest Listening R Factor	93		
		Dest Listening MOS Score	4.20		
		Optimal Listening R Factor(MOS Score)	93% (20)		
		Stream Quality Index (SQI)	A+ (100.00)		

Ready: For Help, press F1

Session: 02:00:29 Active Calls: 76 Calls Completed: 72,695 Avg. Call Rate: 9.457/sec Errors Detected: 0 Error Rate: 0.000% Registered: 0

The watch view is designed to provide an in-depth view of each VoIP call and that has been associated with a user-defined “watch” trigger. Each call is represented by an entry, which is updated once every second, in the topmost report. This report contains the following columns:

Call status: The current status of the call. These may be things such as connecting, ringing, connected, error, etc.

Protocol: The values for this field are SIP or H.323.

Found in: This field specifies which call element the value was found in.

Watch trigger: This field specifies what value caused the watch to be triggered.

Started: This is the time (local time) that the call was started.

Duration: The length of time the call is (or was) active.

Terminator: The side that terminated the call (source or destination).

Source Address: The address of the call initiator (caller).

Source ID/E.164: The SIP user ID or H.323 E.164 alias of the caller.

Source Name/H.323 ID: The SIP display name or H.323 ID of the caller.

Destination Address: The address of the call receiver (party called).

Destination ID/E.164: The SIP user ID or H.323 E.164 alias of the party called.

Destination Name/H.323 ID: The SIP display name or H.323 ID of the party called.

Call ID: The SIP or H.323 call ID associated with this call.

Registered With: The gatekeeper's IP address for H.323 calls, or the Proxy's IP address for SIP calls.

Conference ID: The conference ID (H.323 calls only).

Each individual call has the following sub-views:

- Call Summary
- Call Flow
- Call Trace
- Call Metrics
- Audio Summary
- Audio Details
- Audio QoS
- Video Summary
- Video Details
- Data Details
- RTCP Summary
- RTCP XR Summary
- DTMF Summary

To display information about a particular call, select it in the watch list. Whenever a call is selected, it will remain "locked" in the view for as long as you wish to view its details.

Note: For a detailed description of these sub-views please refer back to the 'Active Calls View' section of the manual.

Endpoints View

WinEyeQ (tm) Professional Edition w/ QoS Version 1.7.0 on 120.249.45.10, 120.249.45.11 - Voice and Video Monitoring and Analysis

File Edit Capture Record View Help

Mode: Analyze Network

Network Monitor Active Calls Recent Calls Recent Errors User Alerts User Alarms Use Watches Endpoints Other Audio Other Video Registrations

Status	ID	IP Address	MAC Address	Description	Discovered	Online	Placed	Received	Successful	Failed	Registered With	Bandwidth Used (KB)
Calling	10024	120.249.45.11	0007E9C9BAC	Win323 Version 1.5.0	10:13:43	00:02:10:12	1,485	0	1,484	0		539,995
Receiving	20024	120.249.0.150	009F808E17	Win323 Version 1.5.0	10:13:43	00:02:10:12	0	1,485	1,484	0		539,995
Calling	10020	120.249.45.11	0007E9C9BAC	Win323 Version 1.5.0	10:13:43	00:02:10:12	1,486	0	1,485	0		539,919
Receiving	20020	120.249.0.150	009F808E17	Win323 Version 1.5.0	10:13:43	00:02:10:12	0	1,486	1,485	0		539,919
Calling	10000	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		127,137
Receiving	20000	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		127,137
Calling	10001	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		127,007
Receiving	20001	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		127,007
Calling	10002	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		127,121
Receiving	20002	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		127,121
Calling	10003	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		126,816
Receiving	20003	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		126,816
Calling	10004	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		126,716
Receiving	20004	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		126,716
Calling	10005	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		126,807
Receiving	20005	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		126,807
Calling	10006	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		126,940
Receiving	20006	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		126,940
Calling	10007	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		127,076
Receiving	20007	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		127,076
Calling	10008	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:02	00:02:08:53	1,473	0	1,472	0		127,113
Receiving	20008	120.249.0.150	009F808E17	WinSP2.7.0	10:15:02	00:02:08:53	0	1,473	1,472	0		127,113
Calling	10009	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:03	00:02:08:52	1,473	0	1,472	0		127,118
Receiving	20009	120.249.0.150	009F808E17	WinSP2.7.0	10:15:03	00:02:08:52	0	1,473	1,472	0		127,118
Calling	10010	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:03	00:02:08:52	1,473	0	1,472	0		126,774
Receiving	20010	120.249.0.150	009F808E17	WinSP2.7.0	10:15:03	00:02:08:52	0	1,473	1,472	0		126,774
Calling	10011	120.249.45.10	0007E9C9BAC	WinSP2.7.0	10:15:03	00:02:08:52	1,473	0	1,472	0		126,797

Endpoint Summary and Recent Call History | Top Talkers

	In/Out	Protocol	Result	Remote ID	Remote Address	Started	Ended	Duration	Aud. Codec	MOS Score	R Factor	Aud. Jitter	Aud. Interval	Vid. Codec	Vid. Jitter	Vid.
Total Calls	1,472															
Successful Calls	1,472															
Failed Calls	0															
Inbound																
Total Calls	0															
Cumulative Time	00:00:00															
Outbound																
Total Calls	1,473															
Cumulative Time	02:08:25															

Ready: For Help, press F1

Session: 02:10:14 Active Calls: 72 Calls Completed: 73,093 Avg. Call Rate: 9.450/sec Errors Detected: 0 Error Rate: 0.000% Registered: 0

The Endpoint View shows a list of each endpoint that has participated in a VoIP call during this WinEyeQ session. This view contains the following columns:

Status: Current status of endpoint, Inactive / Calling / Receiving.

ID: The E.164 alias of the endpoint (H.323) or Call ID (SIP)

IP Address: The IP address of the endpoint.

MAC Address: The MAC address of the endpoint.

Description: A readable text description (if available) of the endpoint.

Discovered: This is the time this endpoint was first observed by WinEyeQ.

Online: The length of time this endpoint has been online.

Placed: The number of calls this endpoint has placed.

Received: The number of calls this endpoint has received.

Successful: The number of calls for this endpoint without errors.

Failed: The number of calls for this endpoint with errors.

Registered With: The gatekeeper's IP address, for H.323 calls, or the Proxy's IP address, for SIP calls.

Bandwidth Used (KB): How many bytes of data has been transferred.

The number of endpoints in the list is user configurable via [Edit | Settings | Endpoints](#).

Endpoint Summary and Recent Call History

[illegible]

The Endpoint Summary and Recent Call History view keeps a list of each call the endpoint has either placed or received.

This view contains the following columns:

In/Out: Whether the call was placed (Outbound) or received (Inbound).

Protocol: The protocol used for the call (H.323 or SIP).

Result: Either Success or Fail.

Remote ID: The E.164 alias (H.323) or Call ID (SIP) of the other endpoint.

Remote Address: The IP address of the endpoint.

Started: The time the call started.

Ended: The time the call ended.

Duration: The call duration.

WinEyeQ User's Guide

Audio CODEC: The type of audio codec this endpoint used.

MOS Score: The average MOS score of this endpoint's audio for the call.

R Factor: The average R Factor of this endpoint's audio for the call.

Audio Jitter: The average jitter of this endpoint's audio for the call.

Audio Interval: The average interval of this endpoint's audio for the call.

Video CODEC: The type of video codec this endpoint used.

Video Jitter: The average jitter of this endpoint's video for the call.

Video Interval: The average interval of this endpoint's video for the call.

The number of endpoint histories in the list is user configurable via Edit | Settings | Endpoints.

Top Talker

ID	Description	Total Calls	Total Time	Total Bandwidth Used (KB)
10020	WinSIP/2.4.9	90	00:05:23	0
20020	WinSIP/2.4.9	90	00:05:23	0
20005	Win323 Version 1.5.0 Beta	67	01:06:51	18,104
50005	Win323 Version 1.4.3	67	01:06:51	18,104
20004	Win323 Version 1.5.0 Beta	67	01:06:53	18,126
50004	Win323 Version 1.4.3	67	01:06:53	18,126
20001	Win323 Version 1.5.0 Beta	66	01:06:00	18,188
50001	Win323 Version 1.4.3	66	01:06:00	18,188
20002	Win323 Version 1.5.0 Beta	66	01:05:58	18,167
50002	Win323 Version 1.4.3	66	01:05:58	18,167
20003	Win323 Version 1.5.0 Beta	66	01:05:55	18,149
50003	Win323 Version 1.4.3	66	01:05:55	18,149
20007	Win323 Version 1.5.0 Beta	51	00:50:31	13,995
50007	Win323 Version 1.4.3	51	00:50:31	13,995
20009	Win323 Version 1.5.0 Beta	51	00:50:30	13,977
50009	Win323 Version 1.4.3	51	00:50:30	13,977

This view contains the following columns:

ID: The E.164 alias of the endpoint (H.323) or Call ID (SIP)

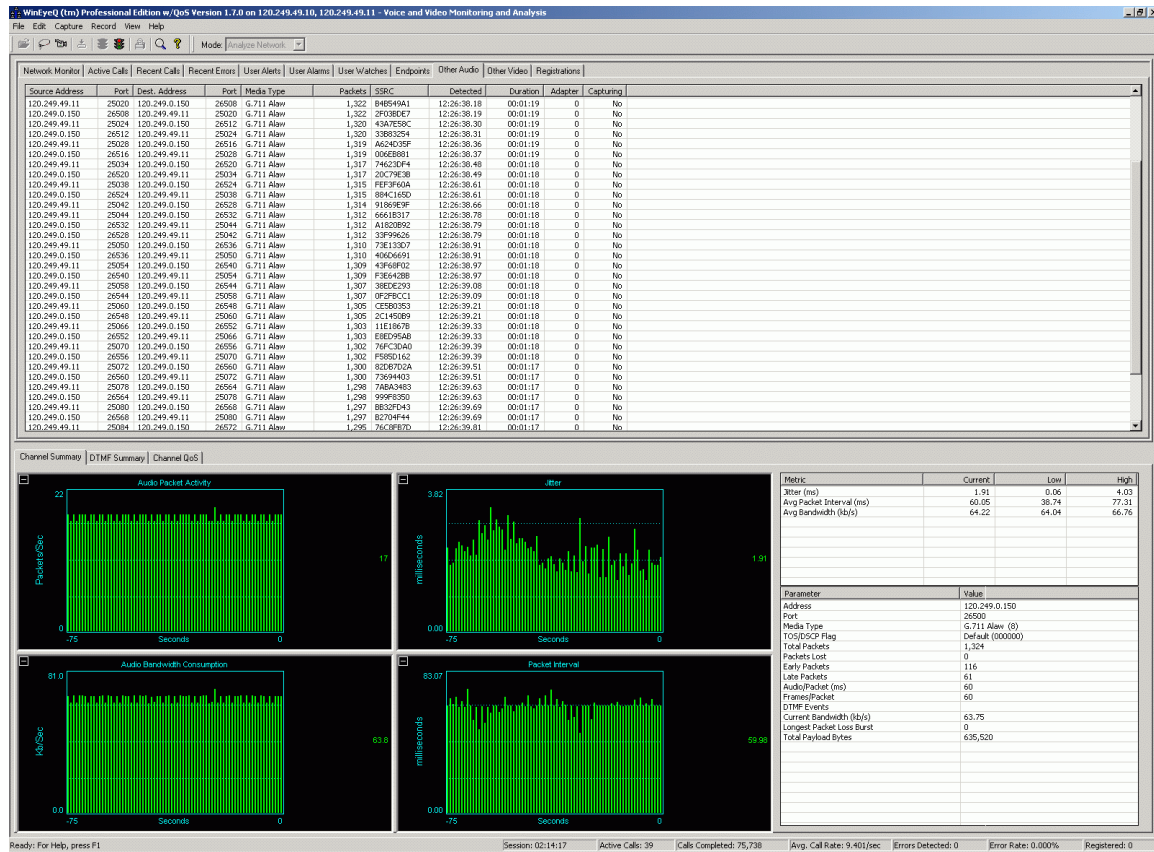
Description: A readable text description (if available) of the endpoint.

Total Calls: The total calls placed and received by this endpoint.

Total Time: The total time the calls lasted.

Total Bandwidth Used (KB): The total bytes transferred.

Other Audio View



This view contains the following columns:

Source Address: The IP address of the call initiator (caller).

Port: The port of the call initiator (caller).

Destination Address: The IP address of the call receiver (party called).

Port: The port of the call receiver (party called).

Media Type: The type of media flowing on this channel.

Packets: The number of packets sent on this channel.

SSRC: The synchronization source from the RTP header.

Detected: The time this stream was detected.

Duration: The length of time the call is (or was) active.

Adapter: The adapter (NIC) that received the packets.

Capturing: Whether or not this stream is being captured to disk.

To start capturing the data from one of these streams right-click the mouse on that stream and select 'Start Rogue Stream Capture'. To stop capturing the data from one of these streams right-click the mouse on that stream and select 'Stop Rogue Stream Capture'.

Other Audio Sub-Views

Channel Summary

This view provides jitter and latency measurements for an audio stream not identified with a VoIP call. Included with the graphical representations of jitter and latency are the high low and current values for each as well as the stream type, the sender's IP address and port, the receiver's IP address and port, the number of packets lost and the DTMF sequences if present within the stream (RFC 2833 section 3 Named Telephony Events).

DTMF Summary

This sub-view provides a detailed and organized tabular display for the active DTMF transmissions that occurs during an audio stream that was not identified with a VoIP call.

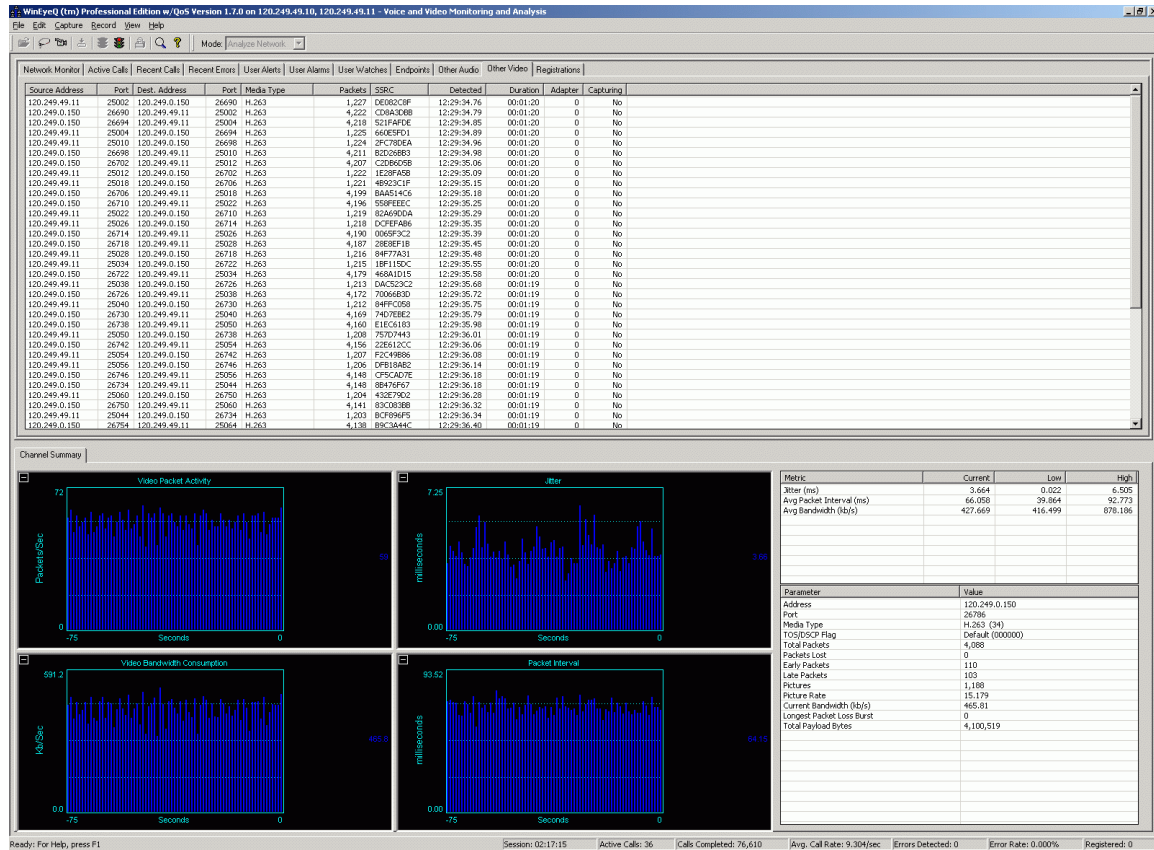
Channel QoS

This sub-view provides the QoS metrics for the stream. The R-factor/MOS scoring feature is a non-intrusive measurement technique available for the WinEyeQ call monitor. WinEyeQ passively measures the characteristics of live audio stream and reports quality scores in real-time. The algorithm used to obtain the R-Factor/MOS quality scores accurately models the way that time-varying impairments, most notably burst packet loss and possible jitter buffer discards, affect perceived speech quality.

Packets	SSRC	Detected	Duration	Adapter	Capturing
226	56DF23BE	09:24:08.25	00:00:04	1	No
258	5BBD4C3A	09:24:08.25	00:00:05	1	No
289	15D33145	09:24:08.25	00:00:05	1	No
328	781F5A71	09:24:08.25	00:00:06	1	No
358	28	Start Rogue Stream Capture	00:00:07	1	No
86	EA	Stop Rogue Stream Capture	00:00:05	1	No
119	A4BC3178	09:24:08.25	00:00:07	1	No
223	4E3B6DF4	09:24:08.27	00:00:04	1	No
257	037D518D	09:24:08.27	00:00:05	1	No

The 'Capturing' column reflects the status of the capture.

Other Video View



This view contains the following columns:

Source Address: The IP address of the call initiator (caller).

Port: The port of the call initiator (caller).

Destination Address: The IP address of the call receiver (party called).

Port: The port of the call receiver (party called).

Media Type: The type of media flowing on this channel.

Packets: The number of packets sent on this channel.

SSRC: The synchronization source from the RTP header.

Detected: The time this stream was detected.

Duration: The length of time the call is (or was) active.

Adapter: The adapter (NIC) that received the packets.

Capturing: Whether or not this stream is being captured to disk

To start capturing the data from one of these streams, right-click the mouse on that stream and select 'Start Rogue Stream Capture'. To stop capturing the data from one of these streams, right-click the mouse on that stream and select 'Stop Rogue Stream Capture'.

Packets	SSRC	Detected	Duration	Adapter	Capturing
226	56DF23BE	09:24:08.25	00:00:04	1	No
258	5BBD4C3A	09:24:08.25	00:00:05	1	No
289	15D33145	09:24:08.25	00:00:05	1	No
328	78455A51	09:24:08.25	00:00:06	1	No
358	28	09:24:08.25	00:00:07	1	No
86	EA	09:24:08.25	00:00:05	1	No
119	A4BC3178	09:24:08.25	00:00:07	1	No
223	4E3B6DF4	09:24:08.27	00:00:04	1	No
257	037D518D	09:24:08.27	00:00:05	1	No

The 'Capturing' column reflects the status of the capture.

Other Video Sub-View

Channel Summary

This view provides jitter and latency measurements for a video stream not identified with a VoIP call. Included with the graphical representations of jitter and latency are the high low and current values for each as well as the stream type, the sender's IP address and port, the receiver's IP address and port, the number of packets lost and the DTMF sequences if present within the stream (RFC 2833 section 3 Named Telephony Events).

Registrations View

The screenshot displays the WinEyeQ Professional Edition interface. The top menu bar includes File, Edit, Capture, Record, View, and Help. Below the menu is a toolbar with icons for various functions. The main window is divided into several tabs: Network Monitor, Active Calls, Recent Calls, Recent Errors, User Alerts, User Alarms, User Watches, Endpoints, Other Audio, Other Video, and Registrations. The Registrations tab is active, showing a table with the following columns: Status, User ID/E.164, User Name/H.323 ID, Address, Registrar/Gatekeeper, Time, TTL, Expires, and Remaining.

Status	User ID/E.164	User Name/H.323 ID	Address	Registrar/Gatekeeper	Time	TTL	Expires	Remaining
Unregistered	10008	"WinSIP 9"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10007	"WinSIP 8"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10006	"WinSIP 7"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10005	"WinSIP 6"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10004	"WinSIP 5"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10003	"WinSIP 4"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10002	"WinSIP 3"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10001	"WinSIP 2"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10000	"WinSIP 1"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10009	"WinSIP 10"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10008	"WinSIP 9"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10007	"WinSIP 8"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10006	"WinSIP 7"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10005	"WinSIP 6"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10004	"WinSIP 5"	120.249.0.135	120.249.0.150	09:57:39			
Unregistered	10003	"WinSIP 4"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10002	"WinSIP 3"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10001	"WinSIP 2"	120.249.0.135	120.249.0.150	09:57:38			
Unregistered	10000	"WinSIP 1"	120.249.0.135	120.249.0.150	09:57:38			
Registering	20009	"Device 10"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20008	"Device 9"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20007	"Device 8"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20006	"Device 7"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20005	"Device 6"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20004	"Device 5"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20003	"Device 4"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		
Registering	20002	"Device 3"	120.249.0.111	120.249.0.150	09:57:33	01:00:00		

Below the table, there are three tabs: Registration Flow, Registration Trace, and Registration Info. The Registration Trace tab is active, showing a diagram of a registration flow between two IP addresses: 120.249.0.135 and 120.249.0.150. The flow is labeled "REGISTER Request" and "REGISTER 200 OK". To the right of the diagram, there is a text box containing the SIP registration details:

```
REGISTER sip:120.249.0.150;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 120.249.0.135;branch=29b04b8d5c9a84e272232c
Max-Forwards: 70
From: "WinSIP 10" <asp:10009@120.249.0.150>;tag=2b1c-01dc-e6de-6d
To: "WinSIP 10" <asp:10009@120.249.0.150>
User-Agent: WinSIP/2.6.0
Call-ID: d53a-35859587-0001-Call10
CSeq: 1 REGISTER
Contact: "WinSIP 10" <asp:10009@120.249.0.135:5060;transport=UDP>
Expires: 3600
Content-Length: 0
```

At the bottom of the window, there is a status bar with the following information: Ready: For Help, press F1; Session: 00:04:32; Active Calls: 200; Calls Completed: 200; Avg. Call Rate: 0.741/sec; Errors Detected: 0; Error Rate: 0.000%; Registered: 30.

The registrations view is designed to provide an in-depth view of each VoIP call registration and its status. Each call is represented by an entry, which is updated once every second, in the topmost report. This report contains the following columns:

Status: The current status of the entry. These may be things such as registering, registered, unregistered, etc.

User ID/E.164: The SIP user ID or H.323 E.164 alias of the registered party.

User Name/H.323 ID: The SIP display name or H.323 ID of the registered party.

Address: The address of the registered party.

Registrar/Gatekeeper: The address of the registrar to which the party is registered.

WinEyeQ User's Guide

Time: This is the time of the most recent registration for this party.

TTL: The registration's time-to-live value.

Expires: The time at which this binding expires.

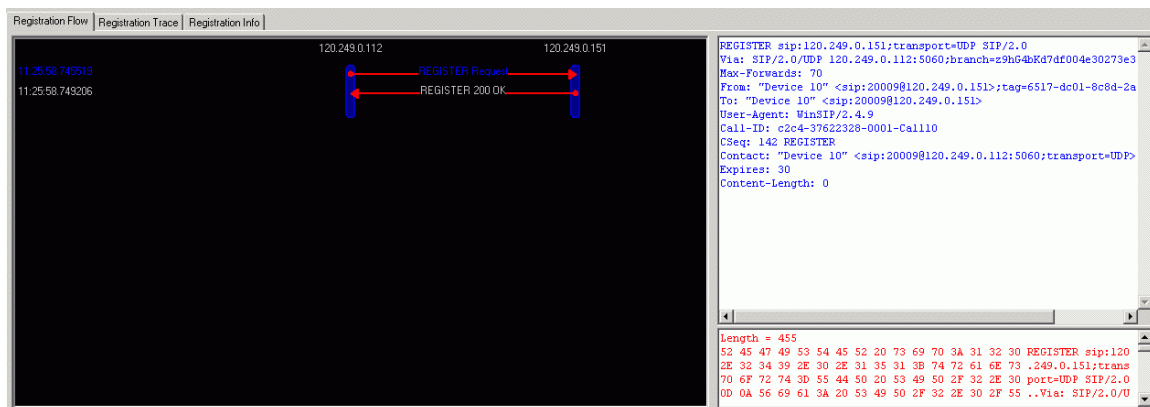
Remaining: The time until this binding expires.

Each individual registration has the following three sub-views:

- Registration Flow
- Registration Trace
- Registration Info

To display information about a particular registration, select it in the registration list. Whenever a registration entry is selected, it will remain “locked” in the view for as long as you wish to view its details.

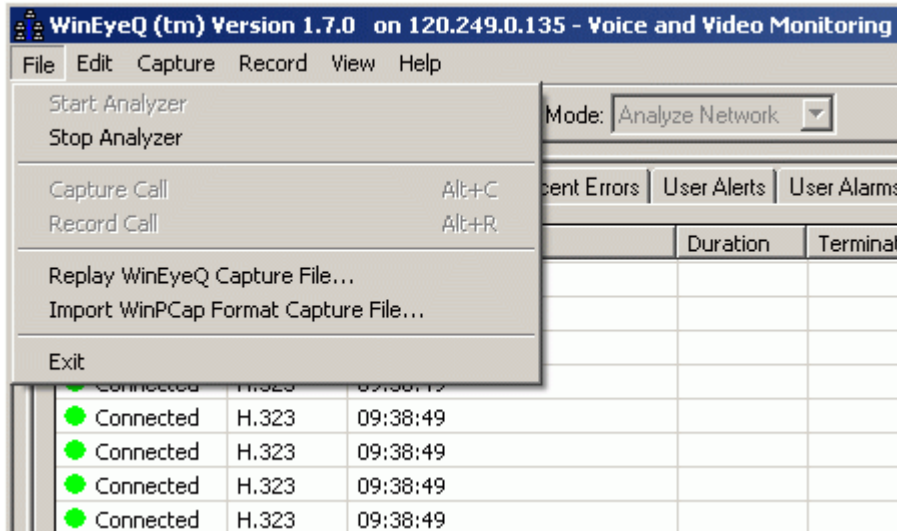
Registration Flow



This view provides a time-stamped ladder diagram view of the registration flow. Each “rung” in the ladder may be highlighted to display the decoded packet in both ASCII and hexadecimal representations.

WinEyeQ Menu Commands

File Menu



This menu contains the commands associated with running WinEyeQ.

Start Analyzer: This command starts the analyzer on the currently selected adapter.

Stop Analyzer: This command stops the current analyzer session.

Capture Call: This command is only enabled when a call is in one of the completed states (completed, error, timeout, etc.). When enabled, this command will capture the selected call in WinEyeQ's proprietary format to the specified file.

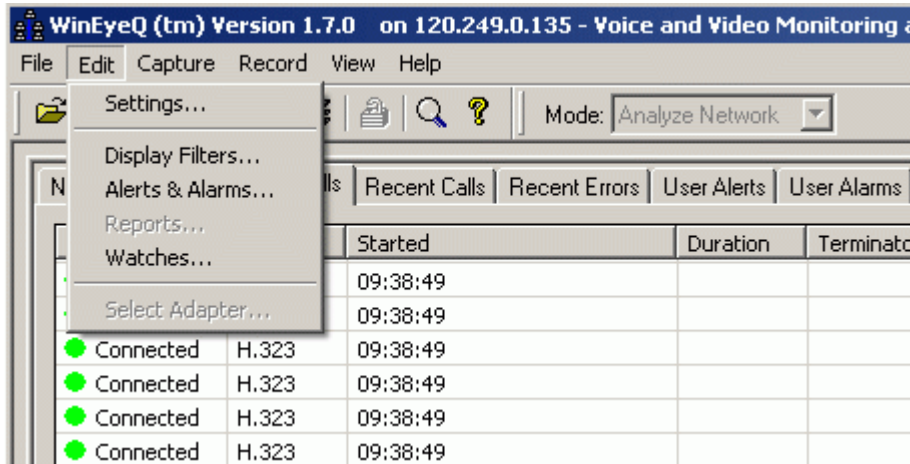
Record Call: This command starts recording the selected call. The signaling and subsequent media will be saved in WinEyeQ's proprietary format to the disk.

Replay WinEyeQ Capture File: This command loads a file captured in WinEyeQ's format and replays it.

Import WinPCap Format Capture File: This command loads a file captured in WinPCap's format and replays it.

Exit: This command ends the WinEyeQ session.

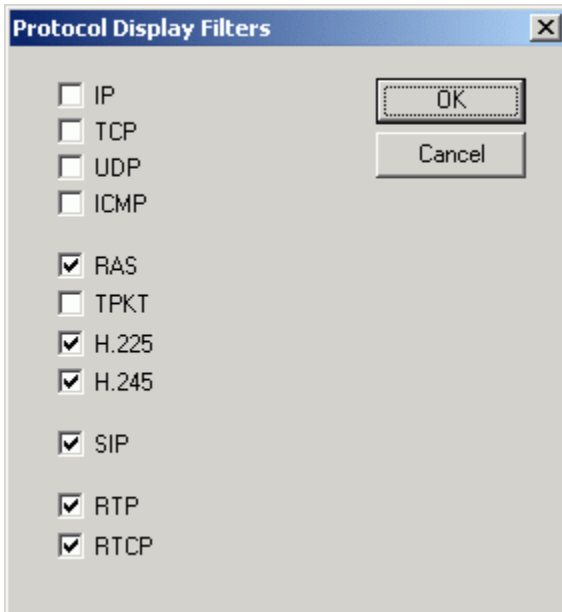
Edit Menu



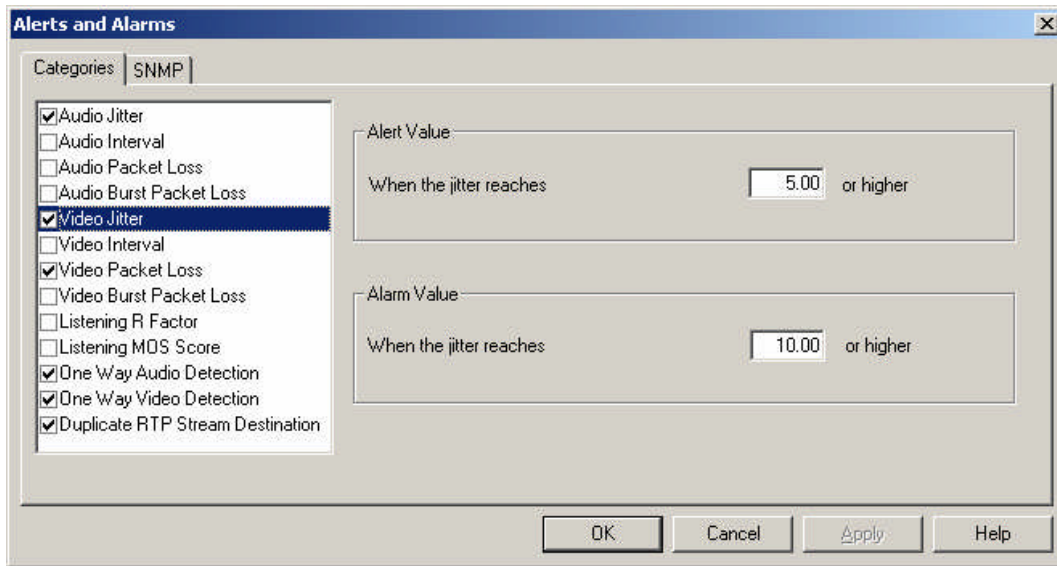
This menu allows you to configure settings, display filters, alerts, alarms, reports, watches, and adapters.

Settings: This command allows you to set the various settings of WinEyeQ so that you can program the tool. See [Configuration Settings](#) for a detailed description.

Display Filters: This command allows you to select the protocol displayed.



Alerts and Alarms: This command allows you to set the Alerts and Alarms that WinEyeQ uses.



The categories of alerts and alarms are:

- Audio Jitter
- Audio Interval
- Audio Packet Loss
- Audio Burst Packet Loss
- Video Jitter
- Video Interval
- Video Packet Loss
- Video Burst Packet Loss
- Listening R Factor
- Listening MOS Score
- One Way Audio Detection
- One Way Video Detection
- Duplicate RTP Stream Detection

Note: For expanded information on Alerts and Alarms, along with SNMP settings see the Alerts and Alarms section of this manual.

Reports: This command allows you to set logging, call, report and preferences settings. See [Configuration Settings](#) for a detailed description.

Watches: From this menu you can add single or multiple watches to WinEyeQ. Watches are a stimulus that triggers WinEyeQ to isolate and analyze any VoIP call that contains that watch. Watches are an extremely simple but powerful way of sifting through a 'haystack' of calls to find the 'needle' call that you are looking for. Calls that are found this way are added to the Watch View.

The screenshot shows the 'Watches' dialog box in WinEyeQ. It features a title bar with a close button. The main area is divided into two sections: 'Add/Edit Single Watch' and 'Add Multiple Watches'. The 'Add/Edit Single Watch' section includes a text box for 'Value' containing '120.249.0.136' and an 'Add' button. The 'Add Multiple Watches' section includes 'From' and 'To' text boxes and an 'Add' button. To the right of these sections are buttons for 'OK', 'Cancel', 'Reset', 'Edit', 'Delete', and 'Clear'. Below these sections is a list box containing the following text: '120.249.0.135', '100001', '100002', '100003', '100004', '100005', '100006', '100007', '100008', '100009', '100010', and 'John Doe'. To the right of the list box are checkboxes for 'Persistent' (checked) and 'Case Sensitive' (unchecked). At the bottom right is a 'Term Matching' section with two radio buttons: 'Exact Phrase' (selected) and 'Match Pattern'.

Add/Edit Single Watch: This is where a watch value is entered. This value can represent any field of any protocol message that WinEyeQ examines. WinEyeQ currently examines the following message fields:

- Source MAC Address, Destination MAC Address
- Source IP Address, Destination IP Address
- Call ID, Conference ID
- Source URI, Destination URI

WinEyeQ User's Guide

- Registrar address, Gatekeeper address
- Source User ID, Destination User ID
- Source E.164, Destination E.164, Source H.323 ID, Destination H.323 ID
- Calling Party Number, Called Party Number
- Call Reference Value, Q.931 Display Name

All the user must do is to add the text string of the value of the field he is looking for.

Note: No quotes are needed for strings that contain blanks.

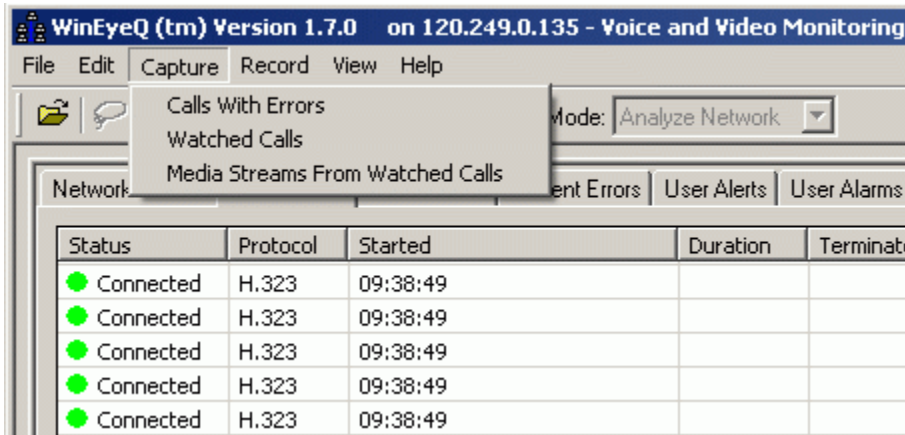
Add Multiple Watches: This is where a range of watches can be added to the program, instead of adding each value separately.

Persistent: If persistent is selected, the watches that have been entered will be written to a file and reloaded the next time that WinEyeQ is run. Otherwise they will be discarded when the program terminates.

Case Sensitive: If case sensitive is selected, the case (upper / lower) of alphabetic characters is considered in the compare. If case sensitive is true then the string "Joe" is not equal "joe".

Term Matching: Exact Phrase or Match Pattern. This offers the user a 'wild card' method of comparing strings. For example, if you add "192.168.10." and have selected Exact Phrase', all fields examined must contain that string exactly. If you have selected Pattern Match, any field that contains "****192.168.10.***" (where * can be any character) will match.

Capture Menu



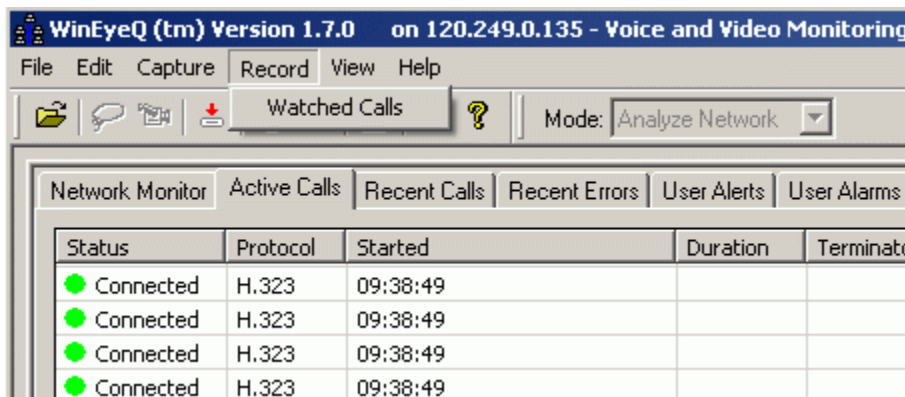
This menu toggles on and off the various capture options.

Calls with errors: Enables/disables capturing calls with errors.

Watched calls: Enables/disables capturing watched calls.

Media streams from watched calls: Enables/disables capturing media streams from the watched calls.

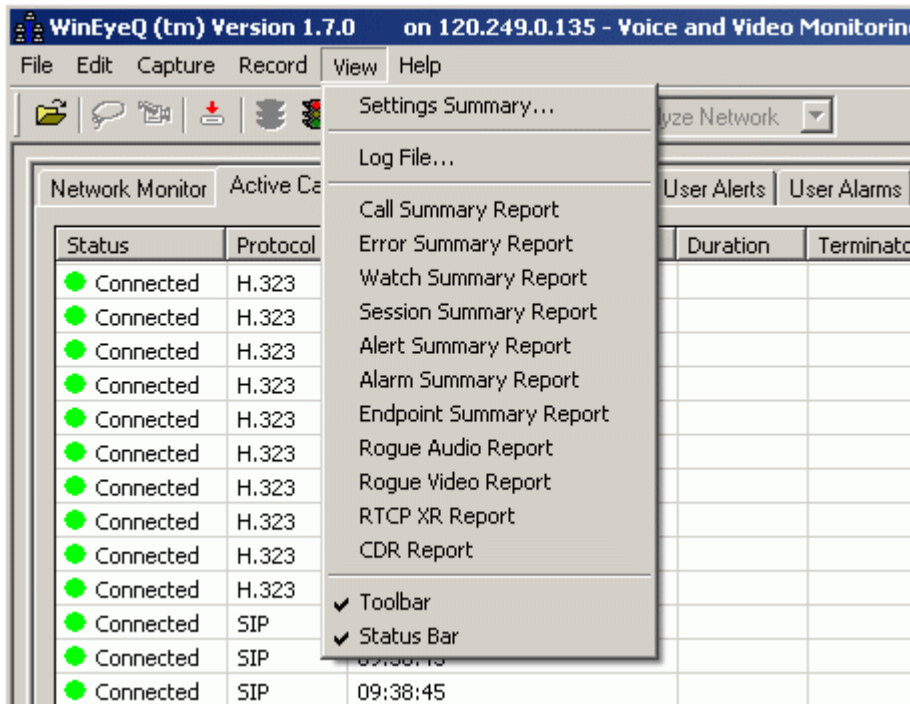
Record Menu



This menu toggles on and off the various record options.

Watched Calls: Enables/disables recording watched calls.

View Menu



This menu allows you to view the settings summary, text based log file and the various reports that are available. It also allows the user to hide the toolbars and status bars.

Settings Summary: Shows the active settings for WinEyeQ.

Log File: Text based data file of the results from the previous test.

Call Summary Report: The call summary report provides a single line entry for each call. Summary information including start time, end time, duration, ID's, addresses, packet counts, QoS metrics, etc. are displayed for each line item.

Error Summary Report: Shows the errors that have occurred during the test session.

Watch Summary Report: Shows the summary information that pertains only to the calls in the watch list.

Session Summary Report: Shows the high level summary information about the test session.

Alert Summary Report: Shows the active alert messages, programmed threshold and measured values.

Alarm Summary Report: Shows the active alarm messages, programmed threshold and measured values.

Endpoint Summary Report: This report contains the information that is removed from the Endpoint View when the number of endpoints in the view exceeds the number of endpoints the user has elected to observe (via Edit | Settings | Endpoints).

Rogue Audio Summary: This report details the start time, end time, duration, QoS measurements, etc. of audio streams that WinEyeQ has detected that are not associated with any VoIP call.

Rogue Video Summary: This report details the start time, end time, duration, QoS measurements, etc. of audio streams that WinEyeQ has detected that are not associated with any VoIP call.

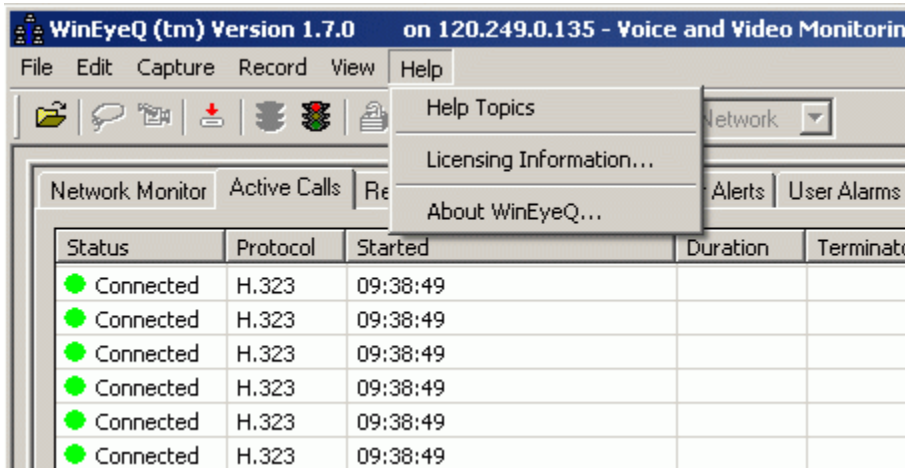
RTCP XR Report: This report captures the information from RTCP XR reports that are sent on the RTCP channel (if any).

CDR Report: Shows the call data records for all the monitored calls.

Toolbar: Shows or hides the toolbar.

Status bar: Shows or hides the status bar.

Help Menu



This menu displays licensing and help information.

Help Topics: Provides user with on line assistance for operating procedures, configuration information and guidance.

Licensing information: Displays information about your WinEyeQ license status. This is also where you can upgrade your license with optional features as they become available.

About WinEyeQ: Displays information about this version of WinEyeQ.

Toolbar Shortcuts



The toolbar contains shortcuts to the most commonly used application commands. The following commands are available:

Replay



Capture



Record



Import WinPCap Capture File



Start Analysis



Stop Analysis



Unlock display (de-select currently selected item)



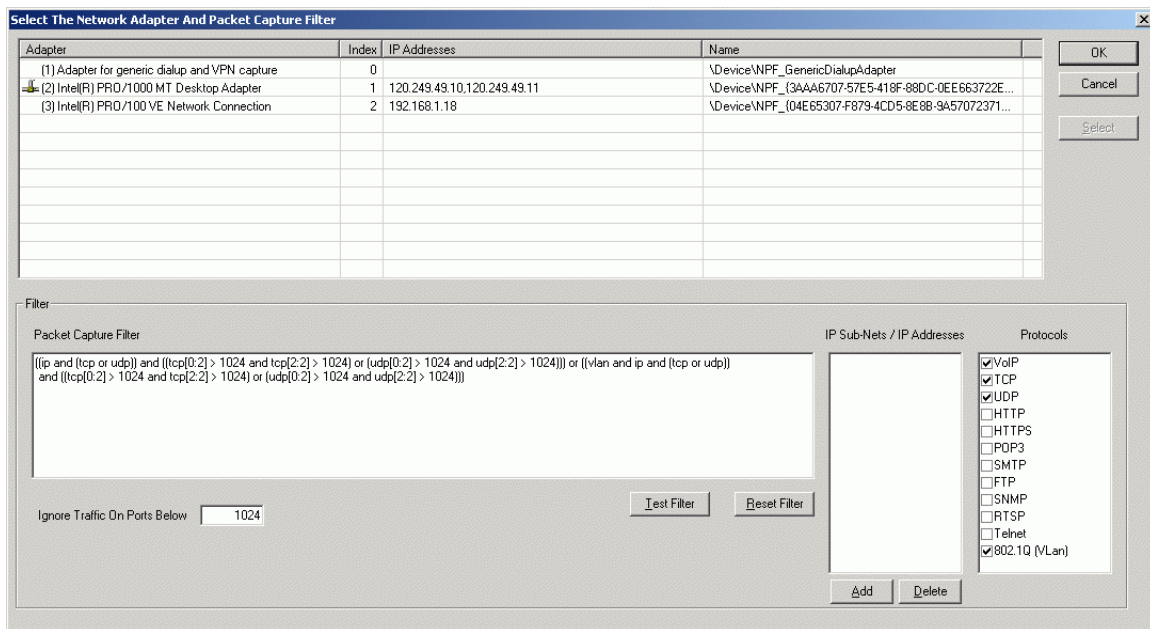
Manage Watches



Help/About



Selecting the Network Adapter and Packet Capture Filter



The first step in preparing to run WinEyeQ is to select the network adapter you wish to monitor. WinEyeQ will automatically display the Select Adapter screen immediately after starting it for the first time. You may also access this dialog from the Edit | Select Adapter menu item.

The Adapter

On the top part of the screen is a list of the Network Adapters that WinEyeQ has discovered on your PC. Select the adapter you want to monitor by clicking the adapter line and then pressing 'Select' or by just double clicking the adapter line.

The Filter

The bottom part of the screen is for the Filter. The Filter is used by the network driver (WinPCap) to decide which packets to send to WinEyeQ and which ones to discard. There are four areas that are used set the Filter, The Packet Capture Filter textbox, The Sub-Nets / Addresses textbox, the Protocols box and the Ports textbox.

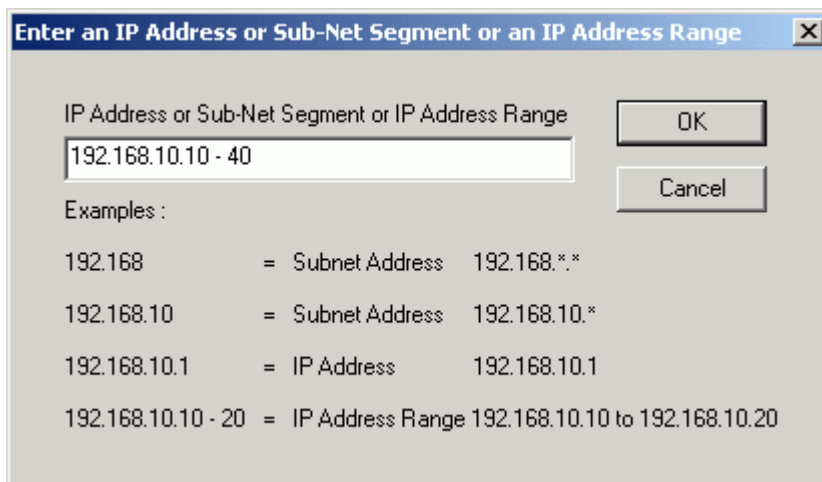
The Packet Capture Filter textbox is the actual Packet Capture Filter. It has been predefined to capture IP, TCP, and UDP packets from all IP addresses with port numbers greater than 1024 on normal and VLAN networks. You may change the Packet Capture Filter by editing the Filter textbox directly, or in combination with the other three textboxes. In case of an error, simply press the 'Reset Filter' button to start over.

WinEyeQ User's Guide

The Sub-Nets / Addresses textbox allows the user to filter on selected IP Addresses or IP Subnets. Subnets / IP Addresses are added or removed from the filter from here.

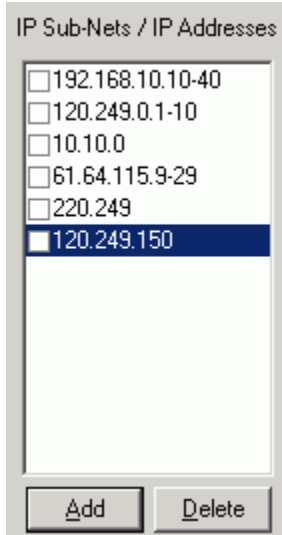


If you click the 'Add' button, the following dialog is displayed:

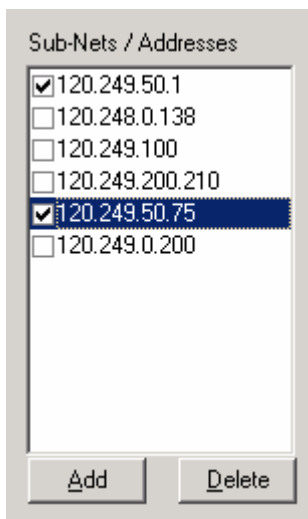


Then enter an IP subnet address, or enter an IP address (or range of addresses), then click OK.

The new value is added to the list.

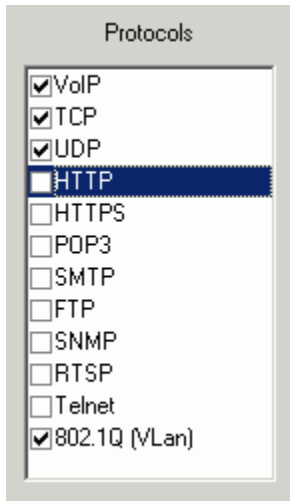


You may use the check boxes to select / deselect the IP addresses you want WinEyeQ to monitor:



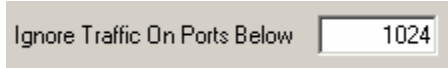
The Subnets / IP Addresses will be added to or removed from the Packet Capture Filter.

The Protocols textbox allows the user to selectively monitor VoIP and other network protocols.

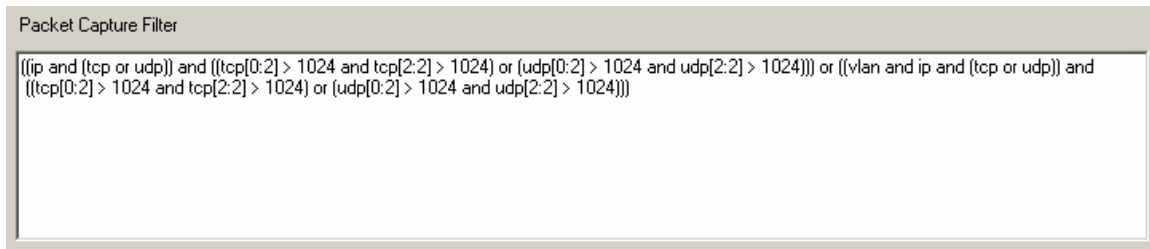


By checking or un-checking these boxes, the indicated protocols are added or removed from the Packet Capture Filter.

The Ports textbox allows the user to selectively exclude packets from a range of port numbers.



The Packet Capture Filter textbox shows the combination of the Subnet / Addresses textbox, the Protocols textbox and the Ports textbox.



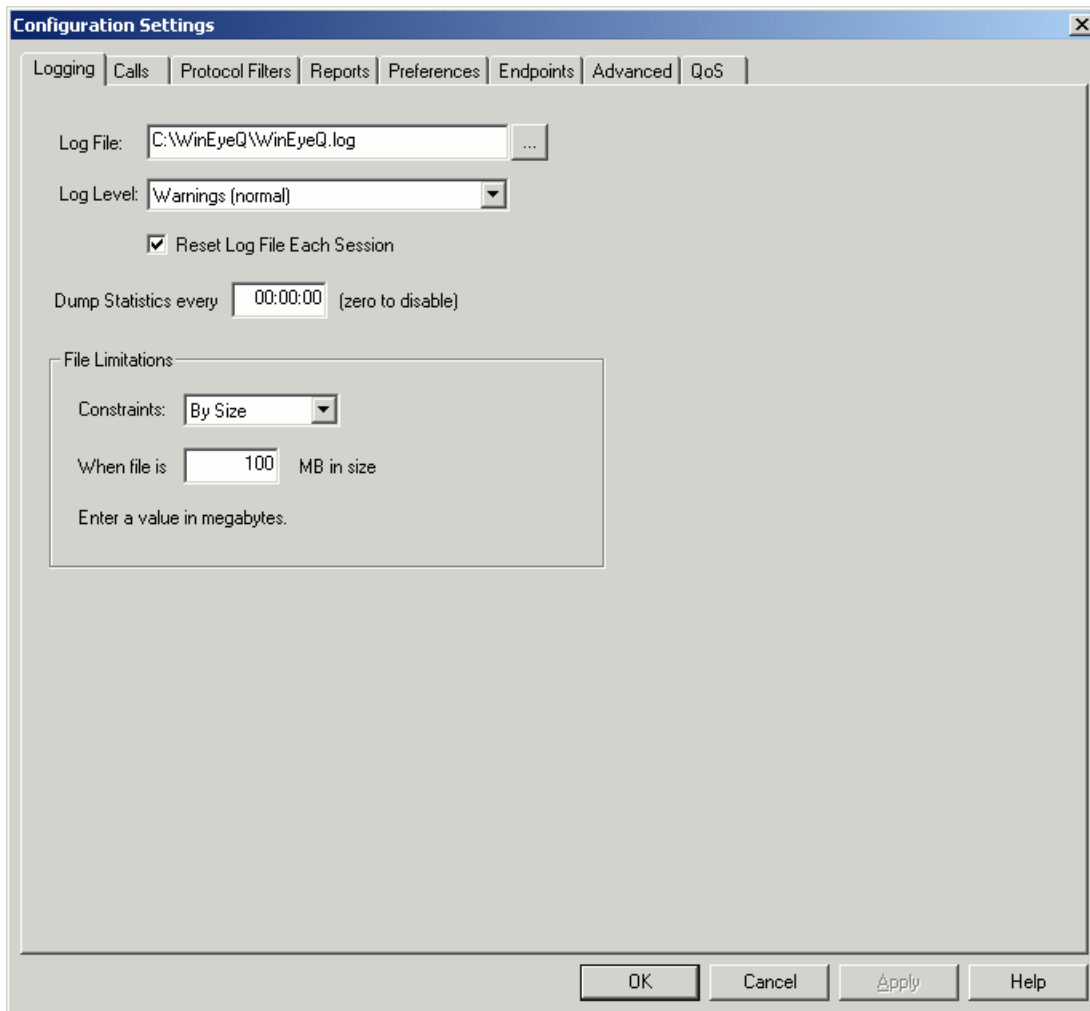
When you make changes to the Subnet /Addresses textbox, the Protocol textbox, or the Ports textbox, the Packet Capture Filter is automatically recalculated. To ensure that the filter has the correct syntax, you may press the Test Filter to check it.

Note: When you press the OK button on the Select Network Adapters dialog, the filter is always checked to ensure it is syntactically correct. If it is not correct, an error message is displayed.

Configuration Settings

The second step in preparing to run WinEyeQ is to review the settings. WinEyeQ will display the following screen(s) when the Edit | Settings menu item is chosen.

Logging



The following options are available to control the application's logging:

Log file: Enter the name and location of the log file you wish to use.

Log level: Select the level of verbosity you wish. The values are:

All: The slowest and most verbose level.

Trace: An extremely high level of detail.

Debug: Standard troubleshooting level.

Information: Medium verbosity.

Warnings: Only warnings and errors.

Errors: Errors messages only.

Reset log file each session: This feature keeps the log file constrained by resetting it after each clean exit. If the previous exit was not clean, the contents of the previous session are preserved.

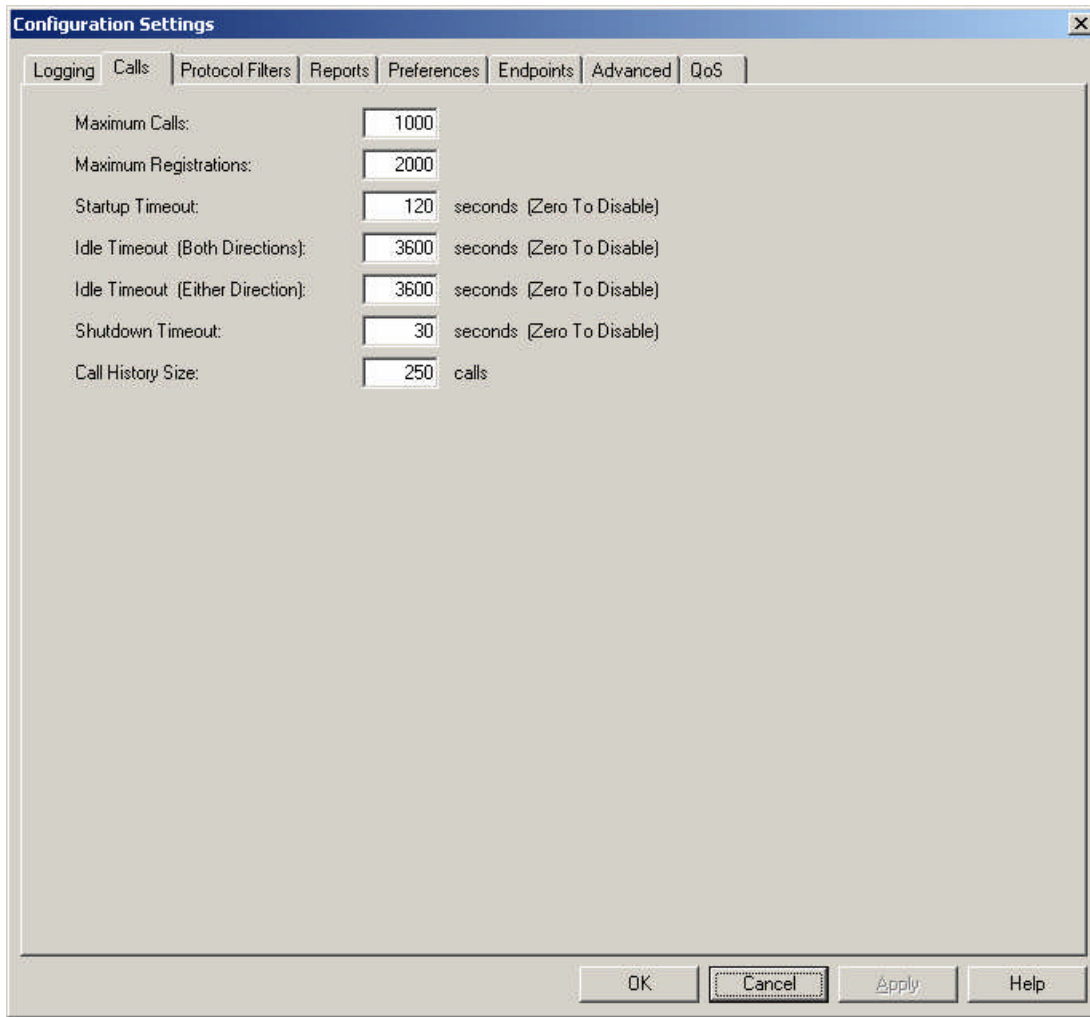
Dump statistics: Sets a timer interval to dump the current statistics to the log file. If this value is zero, the function is disabled.

File Limitations:

Constraints: Sets how the log file is separated. At a certain point the program will close one log file and open a new one and start recording there. The trigger for this event can be set to Size, Interval, Time of Day, or None (which, if selected will hold all information in only one log file).

Constraint Range: Based on the log file constraints, the range sets the event trigger for when the file obtains the value specified in this field.

Calls



The following settings govern the way calls are handled:

Maximum Calls: The maximum number of calls that WinEyeQ will follow (lower values allow for more in-depth media tracing).

Maximum Registrations: The maximum number of registrations that WinEyeQ will follow (lower values allow for more in-depth media tracing).

Startup Timeout: The maximum time after a call is discovered that WinEyeQ will wait for the other endpoint to respond.

Idle Timeout (Both Directions): The maximum time between packets from both the source and destination that WinEyeQ will wait before timing out the call.

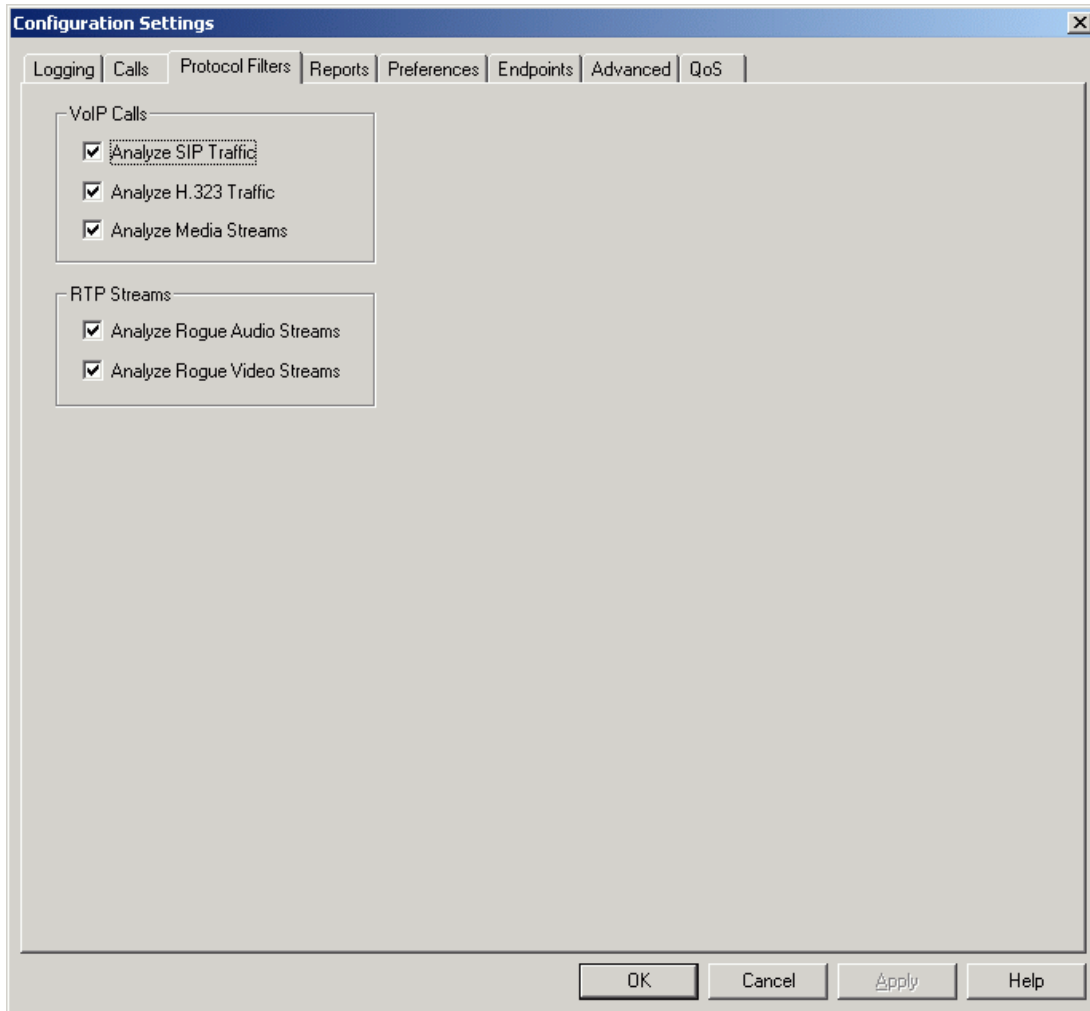
Idle Timeout (Either Direction): The maximum time between packets from either the source or destination that WinEyeQ will wait before timing out the call.

Shutdown Timeout: The maximum time after a call termination attempt is made that WinEyeQ will wait for the other endpoint to respond.

Display time: The time, in seconds, that the call will be displayed on the screen. The amount of memory required by WinEyeQ is proportional to the number of active calls and the length of their display time. Typical values are from 1 to 30 seconds.

Call History Size: The number of calls that will be added to the Recent Calls screen or the Recent Errors screen.

Protocol Filters



The following settings govern the kind of calls are handled:

VoIP Calls:

Analyze SIP Traffic: If checked, WinEyeQ will analyze SIP calls.

Analyze H.323 Traffic: If checked, WinEyeQ will analyze H.323 calls.

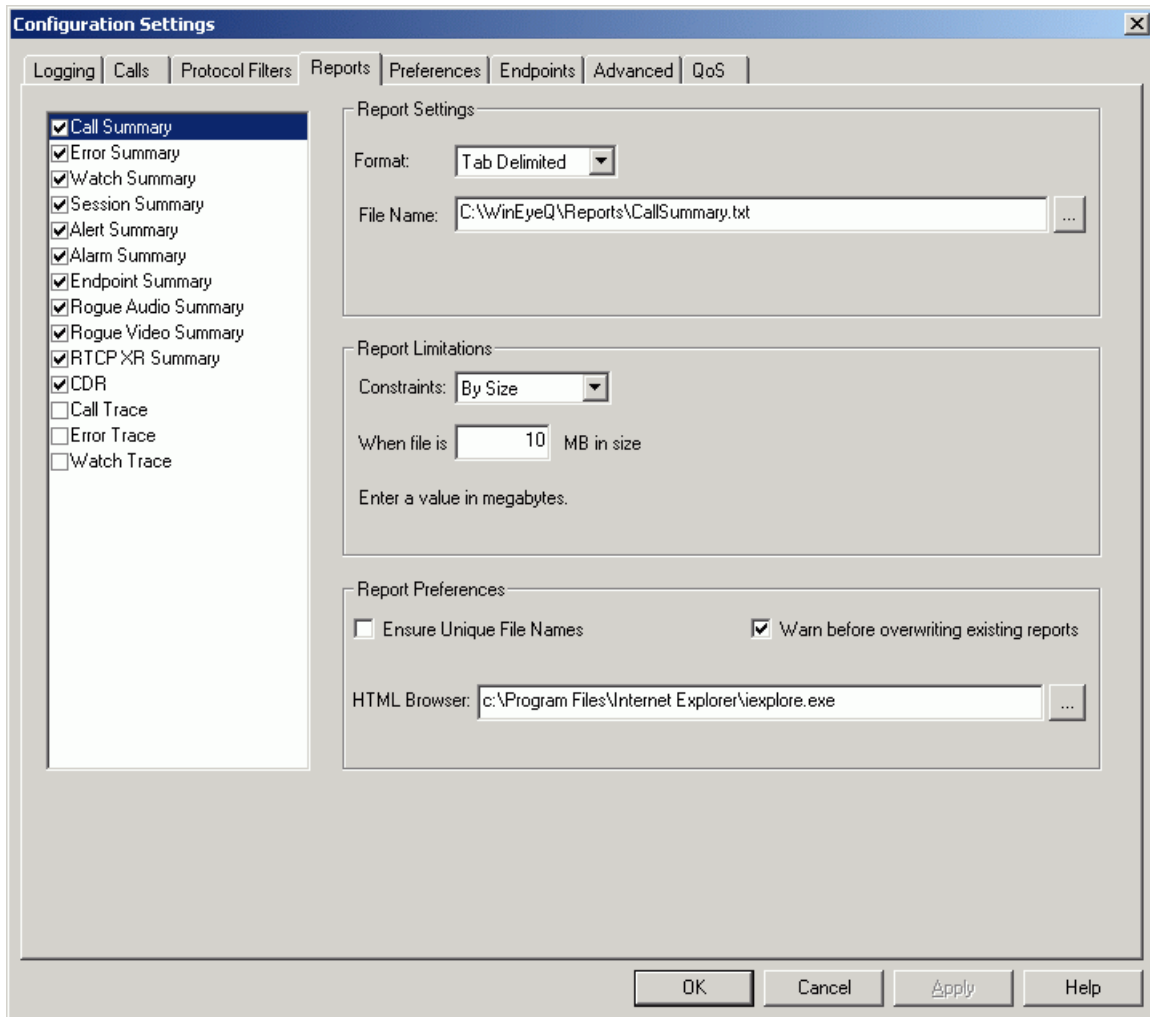
Analyze Media Streams: If checked, WinEyeQ will analyze audio and video streams.

RTP Streams:

Analyze Rogue Audio Streams: If checked, WinEyeQ will analyze audio streams that are not associated with VoIP calls that WinEyeQ is tracking.

Analyze Rogue Video Streams: If checked, WinEyeQ will analyze video streams that are not associated with VoIP calls that WinEyeQ is tracking.

Reports



The following reports are currently available in WinEyeQ:

Call Summary Report: This report has a one-line-per-call format that details the call parameters, start time, end time, duration, QoS measurements, etc. Call Summary reports are saved in the "Reports" folder.

Error Summary Report: This report has a one-line-per-failed-call format that details the call parameters, start time, end time, duration, QoS measurements, etc. Error Summary reports are saved in the "Reports" folder.

Watch Summary Report: This report has a one-line-per-watched-call format that details the call parameters, start time, end time, duration, QoS measurements, etc. Watch Summary reports are saved in the "Reports" folder.

Session Summary Report: This report generates one-line-per-time-interval that details the number of calls passed / failed, network statistics, etc. The 'Time Interval' is set by the user. Session Summary reports are saved in the "Reports" folder.

Alert Summary Report: This report has a one-line-per-alert format that details the call metric, the alert threshold, and the actual value that triggered the alert. Alert Summary reports are saved in the "Reports" folder.

Alarm Summary Report: This report has a one-line-per-alarm format that details the call metric, the alarm threshold, and the actual value that triggered the alarm. Alarm Summary reports are saved in the "Reports" folder.

Endpoint Summary Report: This report contains the information that is removed from the Endpoint View when the number of endpoints in the view exceeds the number of endpoints the user has elected to observe (via Edit | Settings | Endpoints). Endpoint Summary reports are saved in the "Reports" folder.

Rogue Audio Summary Report: This report details the start time, end time, duration, QoS measurements, etc. of audio streams that WinEyeQ has detected that are not associated with any VoIP call. Rogue Audio Summary reports are saved in the "Reports" folder.

Rogue Video Summary Report: This report details the start time, end time, duration, QoS measurements, etc. of video streams that WinEyeQ has detected that are not associated with any VoIP call. Rogue Video Summary reports are saved in the "Reports" folder.

CDR Report: This report has a one line per call format that summarizes the call information. Start time, end time, duration, IP addresses and ID's. CDR Summary reports are saved in the "Reports" folder.

Call, Error and Watch Traces: These reports provide a summary and packet-by-packet trace of the calls. Call, Error and Watch Traces are saved in the "Traces" folder.

Report Settings:

Format: Sets the file format that the report will be rendered in such as ASCII, HTML, or XML.

File Name: Sets the name of the file when it is saved as well as the directory in which it can be found.

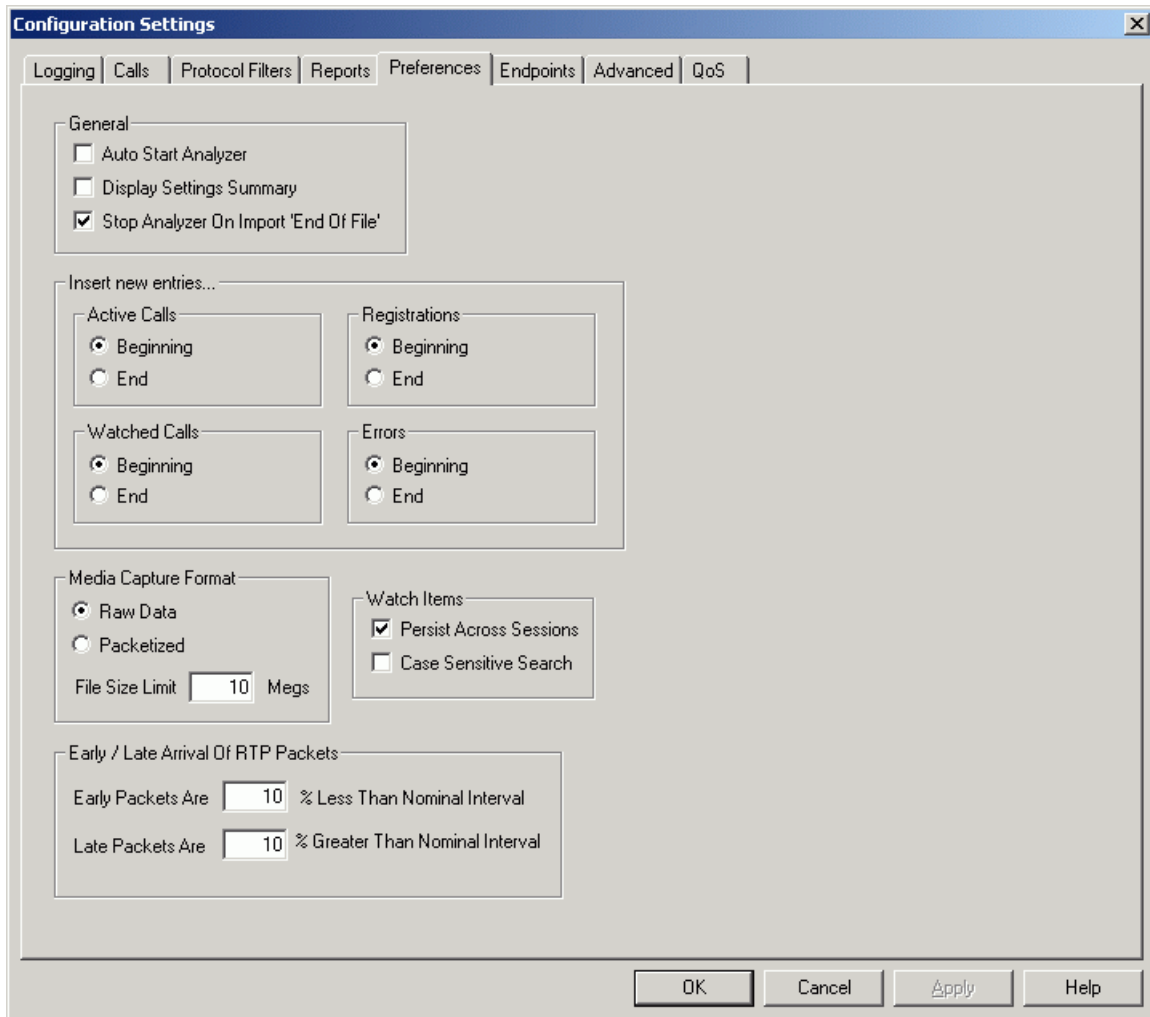
Report Limitations:

Constraints: Sets how each report is separated. At a certain point the program will close one report and open a new one and start recording there. The trigger for this event can be set to Size, Interval, Time of Day, or None (which, if selected will hold all information in only one report file).
Constraint Range: Based on the report constraints, the range sets the event trigger for when the file obtains the value specified in this field.

Report Preferences:

Ensure Unique File Names: If this check box is selected, a timestamp will be appended to the file name when it is created.
Warn before overwriting existing reports: If this check box is selected, the user will be prompted if an existing file is about to be overwritten.
HTML Browser: Specifies the location of an HTML browser application to be used to open reports created in HTML format.

Preferences



The following preferences are available in WinEyeQ:

General:

Auto start analyzer: This feature starts the analysis as soon as WinEyeQ is launched.

Display Settings Summary: If checked, WinEyeQ displays a summary of all the program settings in effect when the program is started.

Stop Analyzer on Import 'End Of File': If selected, WinEyeQ will stop analyzing when an 'end of file' condition is detected while reading an imported file (WinEyeQ or WinPCap format).

Insert new entries:

Active Calls: This option determines where new entries will be added to the active call list.

Watched Calls: This option determines where new entries will be added to the watched call list.

Registrations: This option determines where new entries will be added to the registration list.

Errors: This option determines where new entries will be added to the error list.

Media Capture Format:

Raw or Packetized Data: Choose the media capture format.

File Size Limit: Constraint placed on file size

Watch Items:

Persist Across Sessions: This option automatically reloads the previous session's watches when WinEyeQ is started.

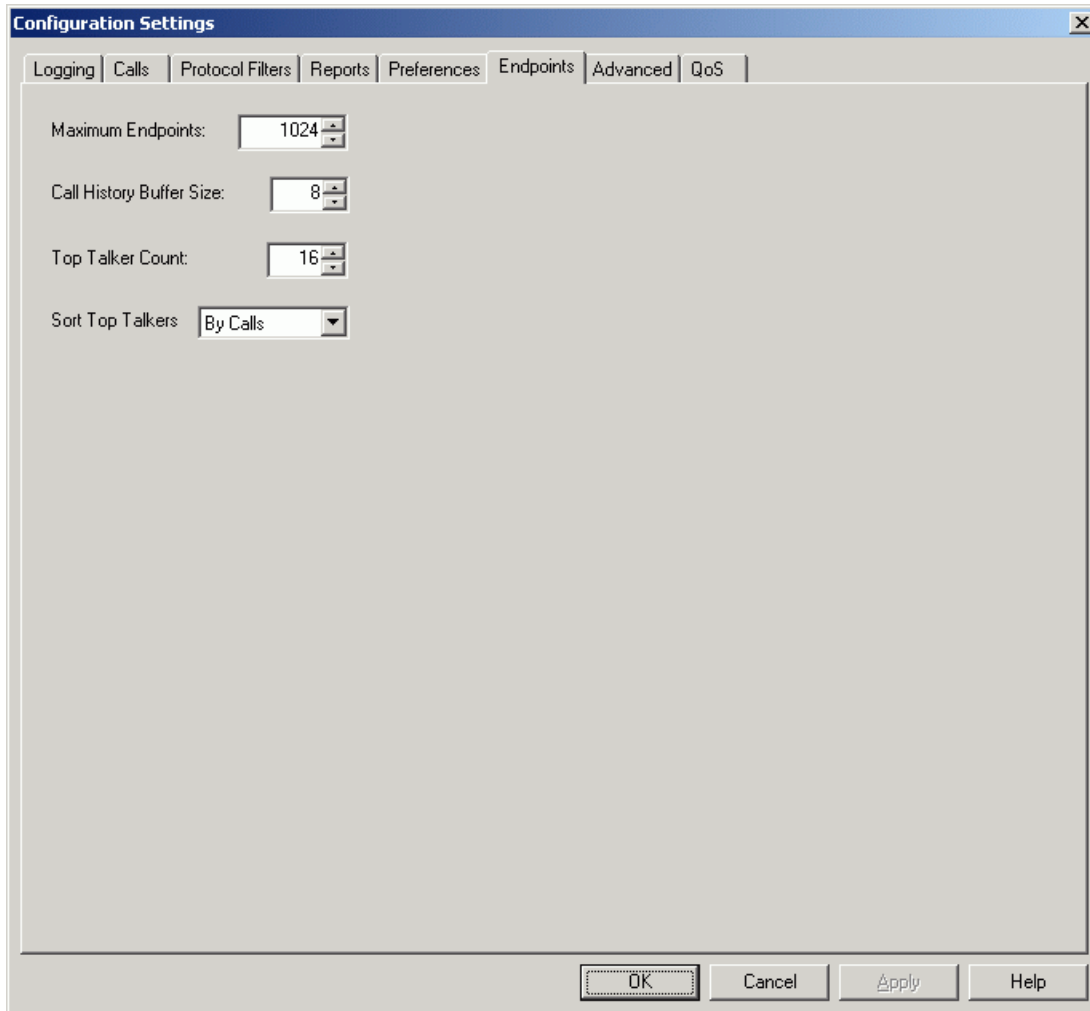
Case Sensitive Searches: This option makes watch item searches sensitive to case.

Early / Late Arrival of RTP Packets:

Early Packets Are XX % Less Than Nominal Interval: Choose a percentage value of the perfect packet interval such that if the interval falls below this value the packet will be considered early. For example, if an RTP stream is sending packets every 20 milliseconds and the early value is 10 %, then if the interval between packets is less than 18 milliseconds, the packet will be considered early.

Late Packets Are XX % Greater Than Nominal Interval: Choose a percentage value of the perfect packet interval such that if the interval falls above this value the packet will be considered late. For example, if an RTP stream is sending packets every 20 milliseconds and the late value is 10 %, then if the interval between packets is greater than 22 milliseconds, the packet will be considered late.

Endpoints



The following endpoint options are available in WinEyeQ:

Maximum Endpoints: The maximum number of endpoints that will be monitored on the Endpoint View.

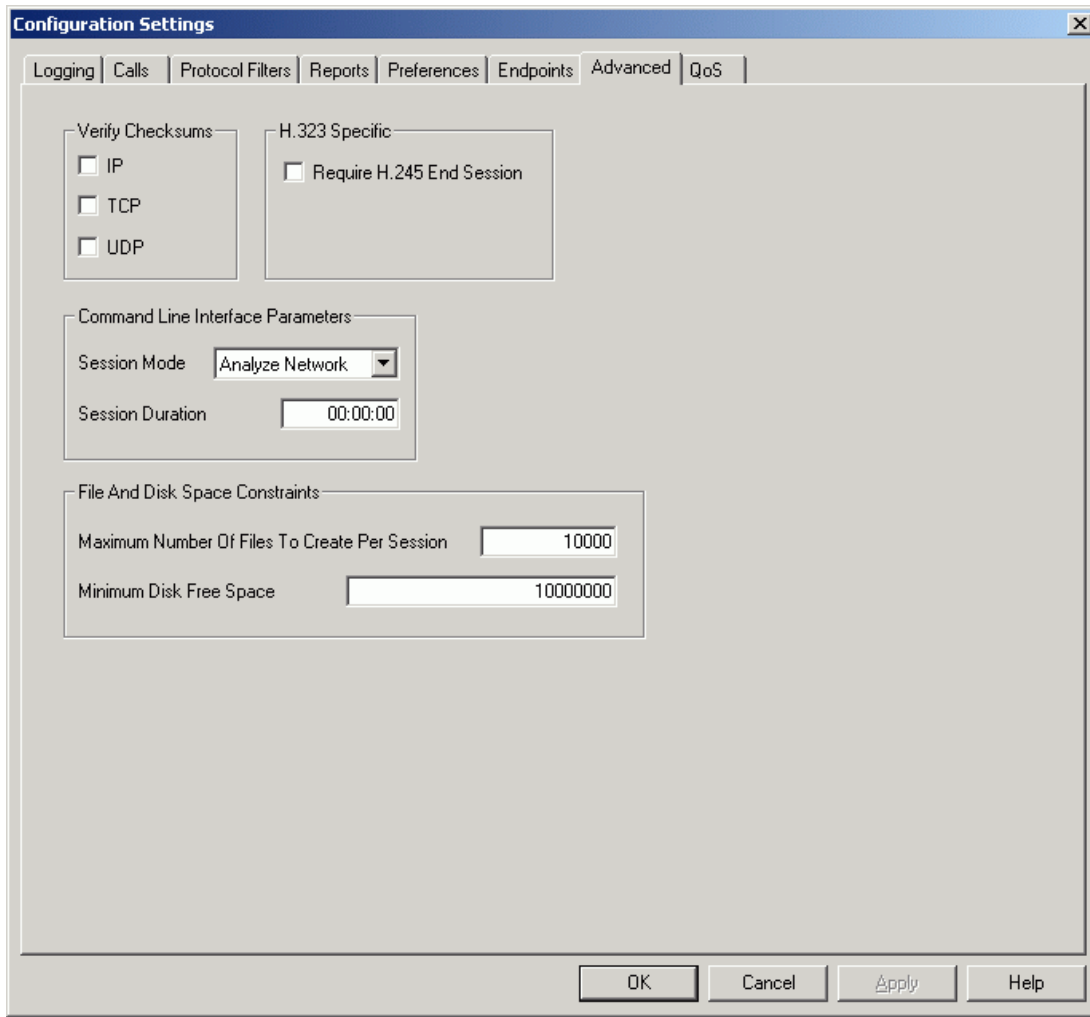
Call History Buffer Size: The maximum number of calls each endpoint has placed / received that will be monitored in the Endpoint Summary and Recent Call History view.

Top Talker Count: The number of Top Talkers that will be added to the Top Talker screen.

Sort Top Talkers: The way that the Top Talkers will be sorted:

- By the number of calls
- By the time those calls were connected
- By the amount of bandwidth used in those calls

Advanced



The following advanced options are available in WinEyeQ:

Verify Checksums:

IP, TCP, and UDP: The respective checksum will be calculated and verified for each of the listed protocols.

H.323 Specific:

Require H.245 End Session: If selected WinEyeQ will require both endpoints to send the End Session message to close the H.245 channel. If not selected, the H.225 Release Complete message will be considered the equivalent.

Command Line Interface Parameters:

Session Mode: Select WinEyeQ mode when program is run by command line.

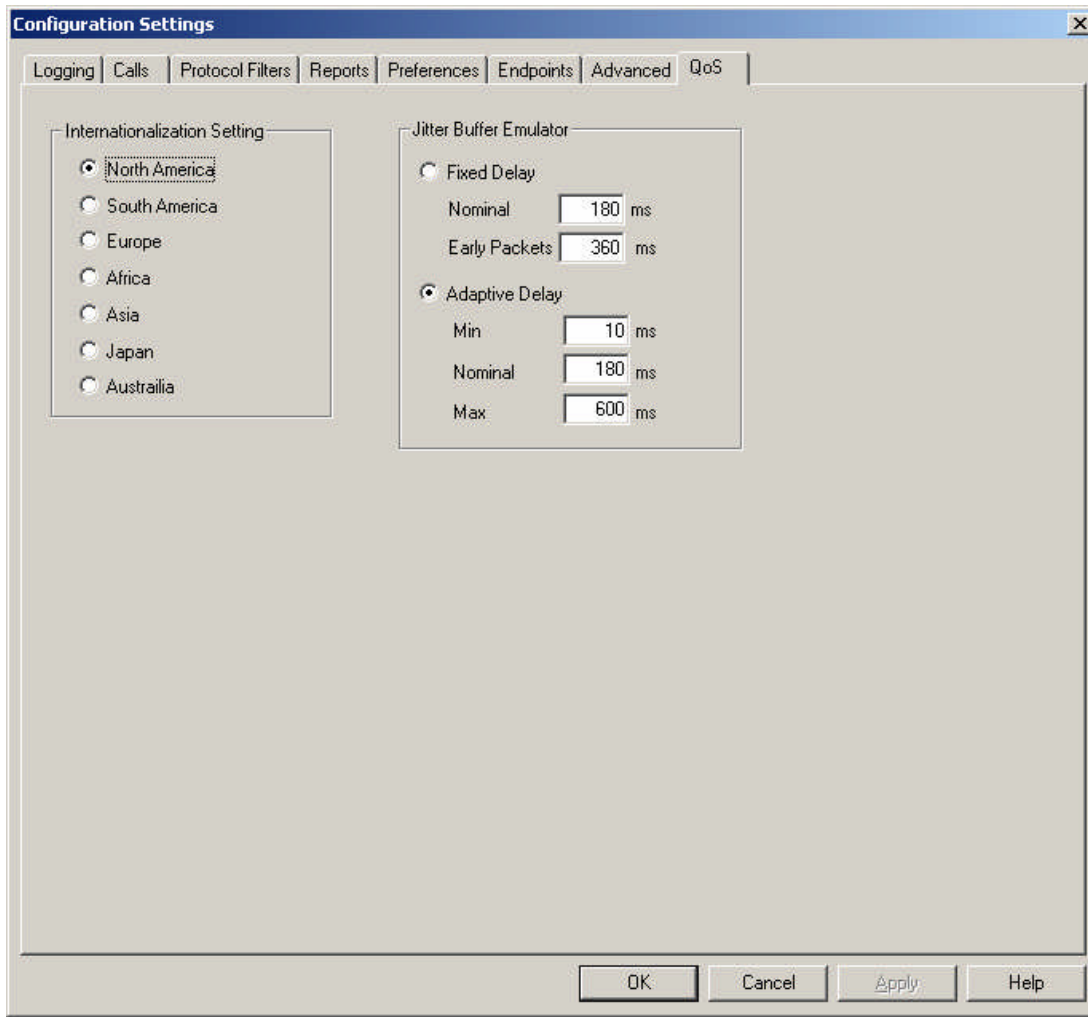
Session Duration: Enter the duration of a command line test. HH:MM:SS

File and Disk Space Constraints:

Maximum Number of Files to Create per Session: This is the maximum number of files (reports, traces, recordings) that WinEyeQ will create during one session.

Minimum Disk Free Space: If the free space on the hard disk that WinEyeQ is running from falls below this amount, WinEyeQ will stop writing to that disk.

QoS



The following QoS options are available for WinEyeQ:

Internationalization Setting: Sets WinEyeQ to generate quality metrics suitable for scales used in different countries.

Jitter Buffer Emulator: simulates the parameters of a jitter buffer. This allows WinEyeQ to have greater accuracy when collecting and analyzing information on packet loss and call quality.

Fixed Delay: Configured Nominal delay does not change.

Nominal Delay: This is the largest “late” delay for a packet beyond which it would be discarded. It is the delay applied to packets that arrives on time or within an “early window”.

Early Packets: This is used to determine whether, a packet that is “early”, can be accommodated in the emulated jitter buffer. It is the maximum delay that will be applied to a packet that is accommodated by the jitter buffer emulator. IP phones and gateways generally have a maximum buffer that limits the total number of packets that can be stored.

The default values for the fixed settings are 60 ms for the Nominal Delay and 80 ms for the Early Packets.

Adaptive Delay: Configured Nominal delay adapts over time.

Min Delay: This is the smallest delay that will be applied to a packet in the Jitter Buffer emulator. The configured nominal delay will not adapt to below this delay.

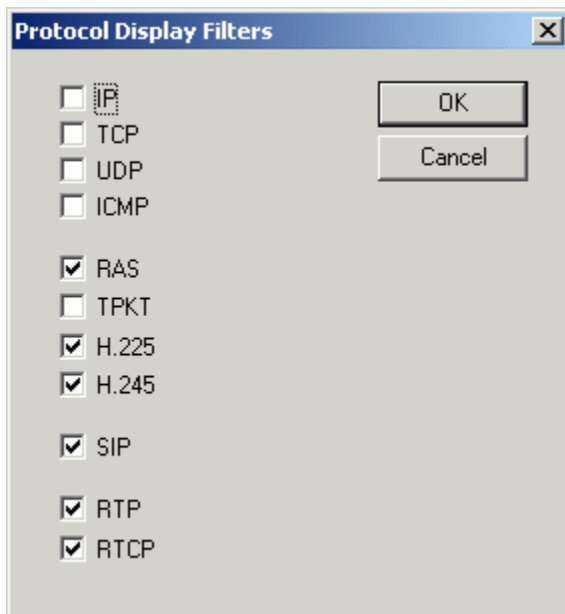
Nominal Delay: This is used as above. Based on the observed jitter, it adapts and is bounded by the Max and Min delays

Max Delay: This is the upper bound for the Nominal delay, and is also used in estimating the “early window” as mentioned above.

The default values for the adaptive settings are 10 ms for the Minimum Delay, 60 ms for the Nominal Delay and 240 ms for the Maximum Delay.

Note: If these values are chosen such that a jitter buffer cannot be constructed, N/C will be displayed in the QoS tables.

Display Filters

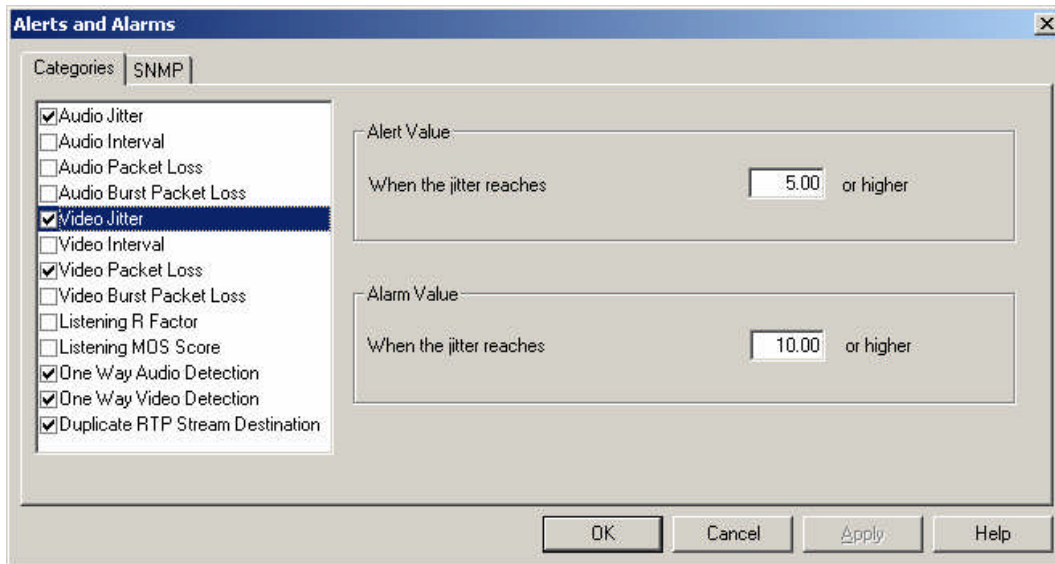


The last step in preparing to run WinEyeQ is to review the filters. WinEyeQ will display the following screen when the Edit | Display Filters menu item is chosen. Select the protocols you want WinEyeQ to display.

Note: Due to memory constraints, only the first few RTP and RTCP packets are displayed for each call.

Alerts and Alarms

Categories



Configurable alerts and alarms are available for the audio and video metric measurements that WinEyeQ performs in real-time. The alert and alarm values are thresholds that are set by the user. The alert and alarm mechanism provides for a two stage detection of user settable limits. Alerts may be set for the following events:

Audio Jitter: When the jitter of an audio stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Audio Interval: When the time between receiving two successive packets (the interval) of an audio stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Audio Packet Loss: When the total number of packets lost of an audio stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Audio Burst Packet Loss: When the number of consecutive packets lost of an audio stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Video Jitter: When the jitter of a video stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Video Interval: When the time between receiving two successive packets (the interval) of a video stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Video Packet Loss: When the total number of packets lost of a video stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Video Burst Packet Loss: When the number of consecutive packets lost of a video stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Listening R Factor: When the listening R factor of an audio stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

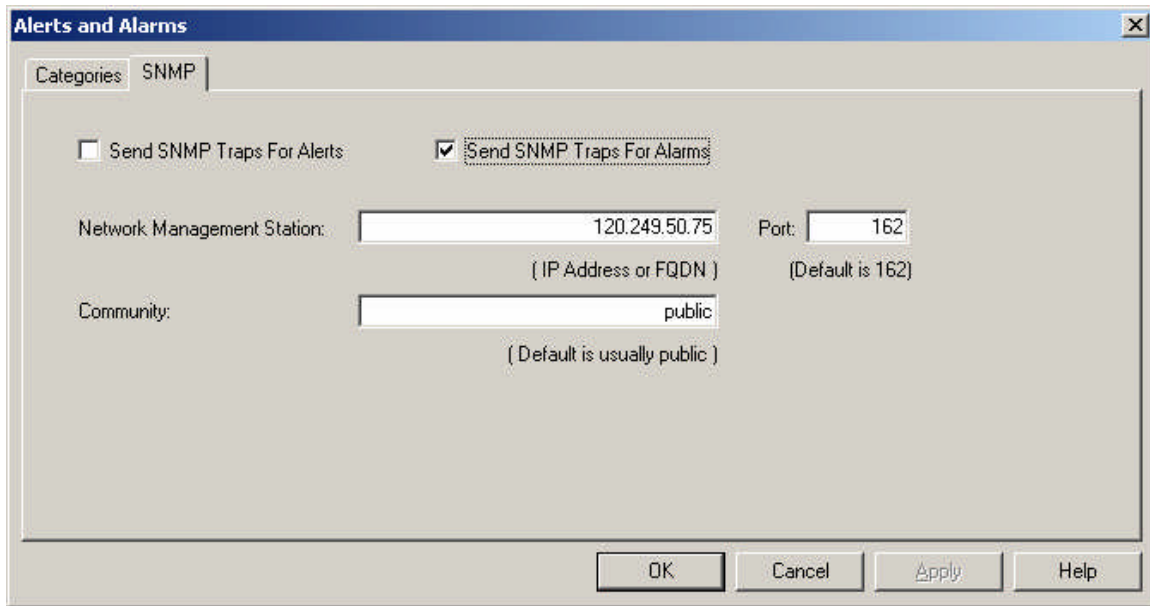
Listening MOS Score: When the listening MOS score of an audio stream exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

One Way Audio Detection: When a call that has audio flowing in only one direction exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

One Way Video Detection: When a call that has video flowing in only one direction exceeds the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

Duplicate RTP Stream Destination: When two media streams that have the same destination IP address and port number are detected and exceed the alert/alarm threshold, a message will be sent to the Alert/Alarm screen.

SNMP



The image shows a Windows-style dialog box titled "Alerts and Alarms" with a close button (X) in the top right corner. Inside the dialog, there is a tabbed interface with two tabs: "Categories" and "SNMP". The "SNMP" tab is currently selected. Below the tabs, there are two checkboxes: "Send SNMP Traps For Alerts" (unchecked) and "Send SNMP Traps For Alarms" (checked). Below these checkboxes, there are three input fields: "Network Management Station:" with the value "120.249.50.75" and a note "(IP Address or FQDN)", "Port:" with the value "162" and a note "(Default is 162)", and "Community:" with the value "public" and a note "(Default is usually public)". At the bottom of the dialog, there are four buttons: "OK", "Cancel", "Apply", and "Help".

Alerts and alarms may optionally send SNMP traps to an SNMP Network Management Station.

Send SNMP Traps For Alerts: If checked an SNMP trap will be sent to the NMS for all Alerts that have been generated by WinEyeQ.

Send SNMP Traps For Alarms: If checked an SNMP trap will be sent to the NMS for all Alarms that have been generated by WinEyeQ.

Network Management Station: This is the IP Address or Fully Qualified Domain Name of the Network Management Station.

Port: This is the port number where the Network Management Station is listening for SNMP traps. The well known port is 162.

Community: This is the community name used by the Network Management Station. The default community name for most SNMP agents is public.

WinEyeQ User's Guide

The following information is provided in each SNMP trap that is sent:

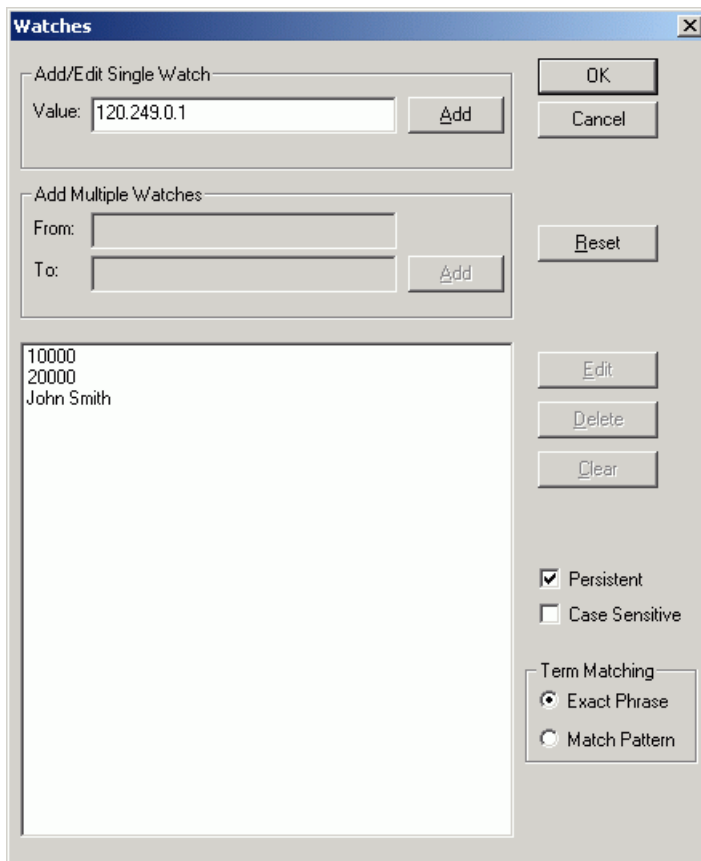
- The name of the application generating the trap - WinEyeQ
- Whether the trap was an alert or an alarm
- The type of trap - audio, video, or signaling
- The description of the trap
- The source IP address and port of the call
- The destination IP address and port of the call
- The threshold value that was set for the trap
- The actual value that triggered the trap
- The date and time the trap was generated
- The ID and User name from the call

Note: Each field of the trap is generated as a character string.

The following is an example of an SNMP trap generated by WinEyeQ:

```
1.3.6.1.4.1.27631.1 WinEyeQ
1.3.6.1.4.1.27631.1.2 Alarm
1.3.6.1.4.1.27631.1.2.2 Video
1.3.6.1.4.1.27631.1.2.2.2 Jitter
1.3.6.1.4.1.27631.1.2.2.2.1 Jitter High Alarm
1.3.6.1.4.1.27631.1.2.2.2.2 Source 120.249.50.100:50354
1.3.6.1.4.1.27631.1.2.2.2.3 Dest 120.249.50.75:25008
1.3.6.1.4.1.27631.1.2.2.2.4 Threshold 1.10
1.3.6.1.4.1.27631.1.2.2.2.5 Value 3.09
1.3.6.1.4.1.27631.1.2.2.2.6 Time 05/08/2007 17:43:16.697
1.3.6.1.4.1.27631.1.2.2.2.7 ID = BB94C810992CEE9D58731FB3D16753F2 -
User = 500095
```

Watches



The watch mechanism allows you to filter out specific calls based upon the value of various call elements or fields within a call. This powerful mechanism allows you to trap calls based upon call ID, IP address, E.164 alias, H.323 ID and most other fields where values are known ahead of time. You may add, edit and delete values associated with watches.

Watches may be designated as persistent (lasting across sessions) and case-sensitive by selecting the appropriate settings on the Edit | Preferences page from the options menu item. Also, you can specify the watch to match the value exactly or match a subset of the value. For example, if 'Exact' were selected, the watch 'Joe' would match the value 'Joe' but not the value 'Joey'. If 'Match Pattern' were selected, 'Joe' would match both 'Joe' and 'Joey'.

Capturing Calls

Overview

WinEyeQ allows you to capture the protocol signaling and initial RTP and RTCP packets from each stream. You may capture the calls manually or associate watches with known call elements and automatically capture the calls as the watches are triggered.

Capturing watched calls also allows you to save the individual media streams with or without the RTP header. This feature is enabled by selecting "Media Streams from Watched Calls" from the Capture menu.

When the file is automatically generated via a watch, the file format will be as follows:

Cap [user id] [timestamp].EyeQ:

Where [user id] is the user ID of the source and timestamp is the local time of the call when recording begins. e.g. "Cap 2156726550 01102005090000.EyeQ".

Capture - Calls with Errors

Click Capture menu command then select Capture Calls with Errors. Calls are captured as they move off the active list to the error list. If you don't have Capture Calls With Errors selected, you can still capture the call by right clicking it in the Error list. Only the initial three RTP packets are captured unless the packet contains DTMF. Only the initial three RTCP packets are captured.

The Default file location where captures are saved is:

C:\WinEyeQ\Capture Files*.EYEQ

Capture - Watched Calls

Click Capture menu command then select Capture Watched Calls. If you don't have Capture Watched Calls selected, you can still capture the call by right clicking it in the Error list. Only the initial three RTP packets are captured unless the packet contains DTMF. Only the initial three RTCP packets are captured

The Default file location where captures are saved is:

C:\WinEyeQ\Capture Files*.EYEQ

Capture - Media Streams from Watched Calls

Click the Capture menu command then select Capture Media Streams from Watch Calls. With this option selected the capture file(s) will contain the full number of RTP packets for the watched call.

If enabled while the analyzer is running media streams from calls already on the watch list will not be captured. Only calls added to the watch list after enabling Capture | Watch Calls Media Streams will have their media streams captured.

Default file location:

C:\WinEyeQ\Audio Capture Files*.g711a (for a call running G.711 Alaw)

C:\WinEyeQ\Video Capture Files*.h261 (for a call running H.261)

Capture - Rogue Streams

While viewing media streams in either Other Audio or Other Video right click on the stream of interest and select Start Rogue Stream Capture.

Default file location:

C:\WinEyeQ\Audio Capture Files*.rogue for Other Audio Capture.

C:\WinEyeQ\Video Capture Files*.rogue for Other Video Capture.

The media stream capture will begin immediately after selecting Start Rogue Stream Capture. Media previous to this point will not be included within the capture file.

Recording Calls

Overview

WinEyeQ allows you to record the protocol signaling, RTP and RTCP channels into one singular file for replay. You may record the calls manually or associate watches with known call elements and automatically capture the calls as the watches are triggered.

When the file is automatically generated via a watch, the file format will be as follows:

Rec [user id] [timestamp].EyeQ:

Where [user id] is the user ID of the source and timestamp is the local time of the call when recording begins, e.g. "Rec 2156726550 01102005090000.EyeQ".

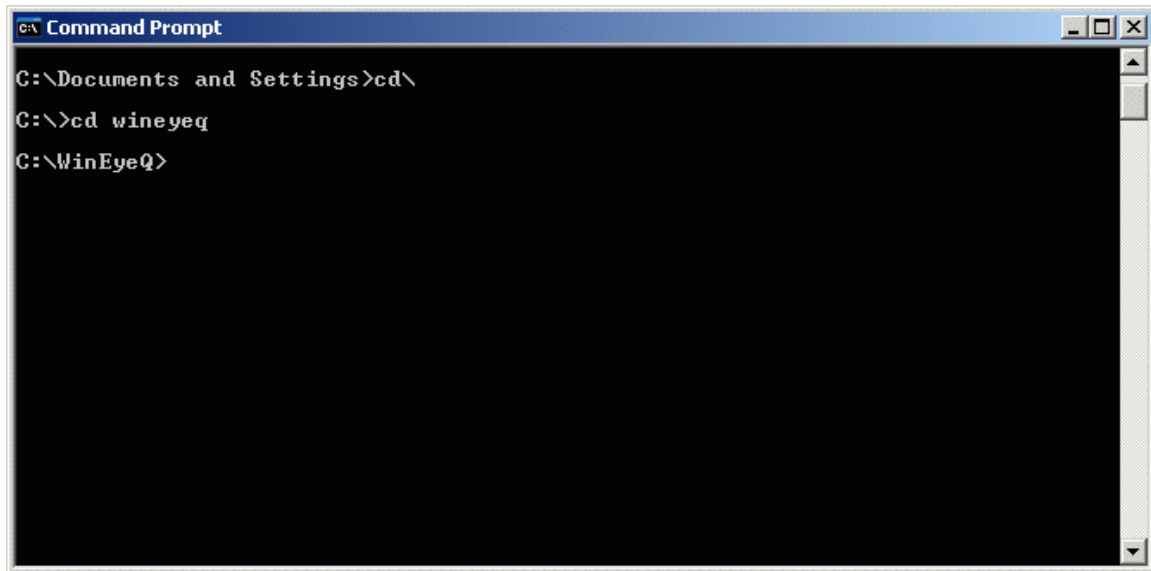
Record - Watch Calls

All of the signaling and media packets are recorded from the watched call to the file.

Command Line Interface

WinEyeQ offers a very basic Command Line Interface (CLI). The application can be started and given all of the information it needs to execute (see Creating Custom Scenarios) but that is the extent of the interaction.

All commands must be entered while in the WinEyeQ directory.



```
CA Command Prompt
C:\Documents and Settings>cd\
C:\>cd wineyeq
C:\WinEyeQ>
```

From here you can access all of the WinEyeQ commands and parameters.

To understand how to use the CLI you must first understand the way WinEyeQ retains the parameters the user has entered from one session to the next. WinEyeQ stores these parameters in files located in the WinEyeQ folder on the installed hard drive.

WinEyeQ.cfg

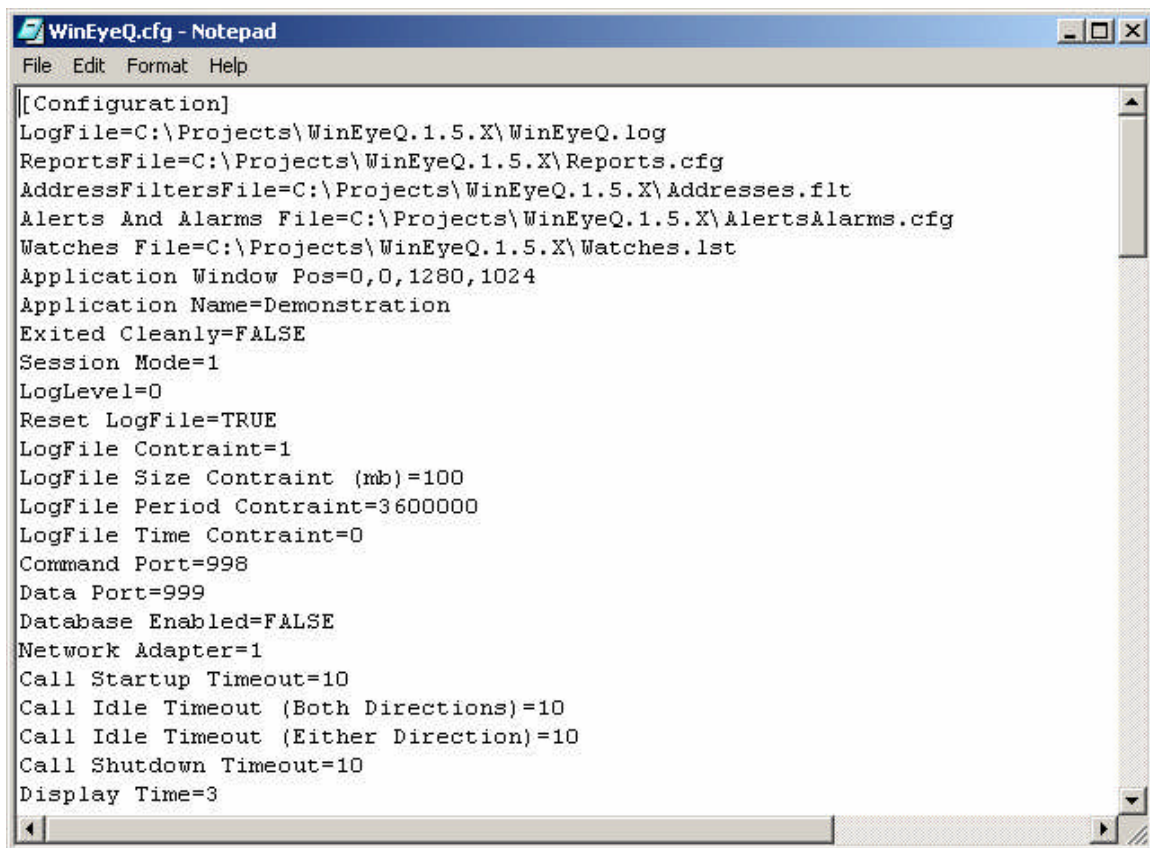
The primary parameter file for WinEyeQ is WinEyeQ.cfg. This file contains the basic parameters required to run WinEyeQ. Normally the values in this file are read and written by the Graphical User Interface (GUI) of the application. If this file is ever deleted, WinEyeQ will rebuild it the next time the application is run.

WinEyeQ.cfg also contains the names of other files that may be used while WinEyeQ is running.

The following parameter files are listed in WinEyeQ.cfg:

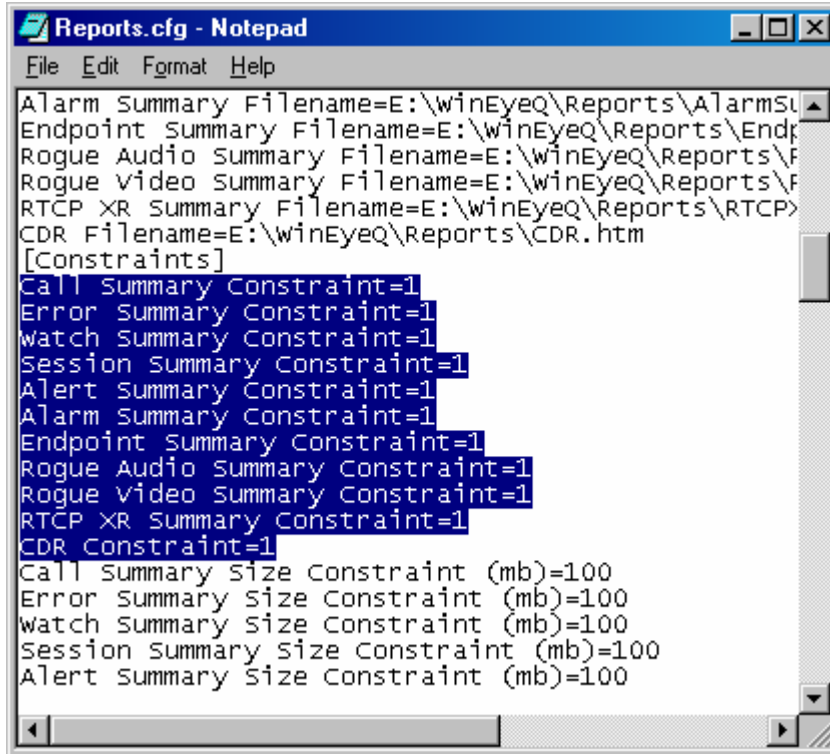
- Reports.cfg
- AlertsAlarms.cfg
- Watches.lst
- Addresses.flt

Each of these files contains information that can be altered to affect the WinEyeQ sessions. Again, the values in these files are normally read and written by the Graphical User Interface (GUI) of the application. To find the location of these files in the configuration, open up WinEyeQ.cfg using Notepad.



Reports.cfg

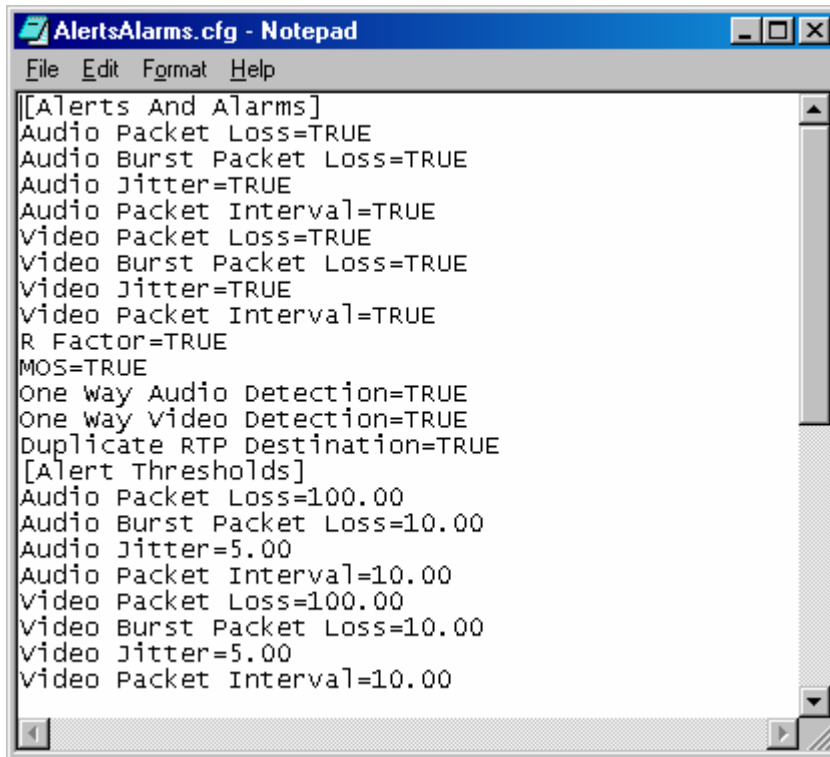
This contains the parameters for the reports that are generated by WinEyeQ. The active reports can be toggled on and off (a "0" is off, while a "1" is on).



You can also find and change the settings on how often the reports are refreshed (such as time of day, file size, or set intervals) as well as various other report constraints.

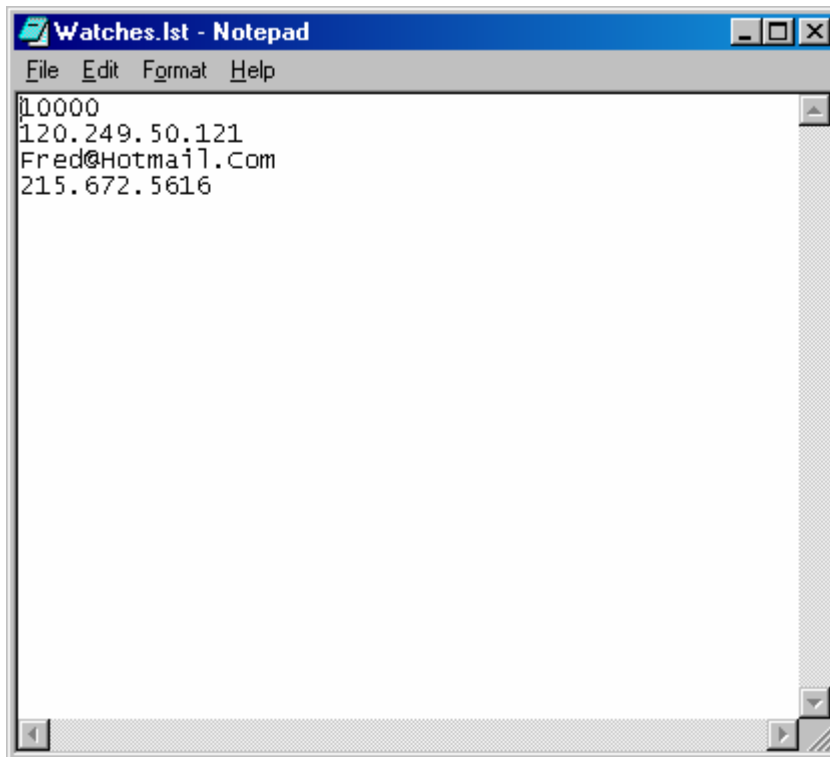
AlertsAlarms.cfg

This file controls which alerts and alarms are active, as well as the various thresholds to trigger them.



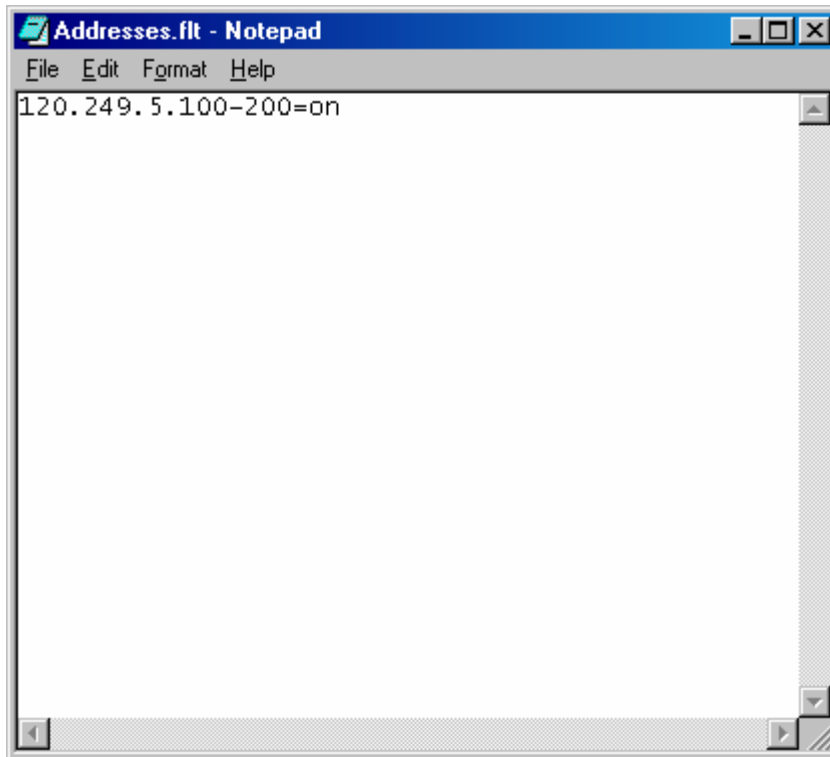
Watches.lst

Sets and controls the watches that are generated.



Addresses.flt

This controls the range of IP addresses that the filter looks at for information to analyze.

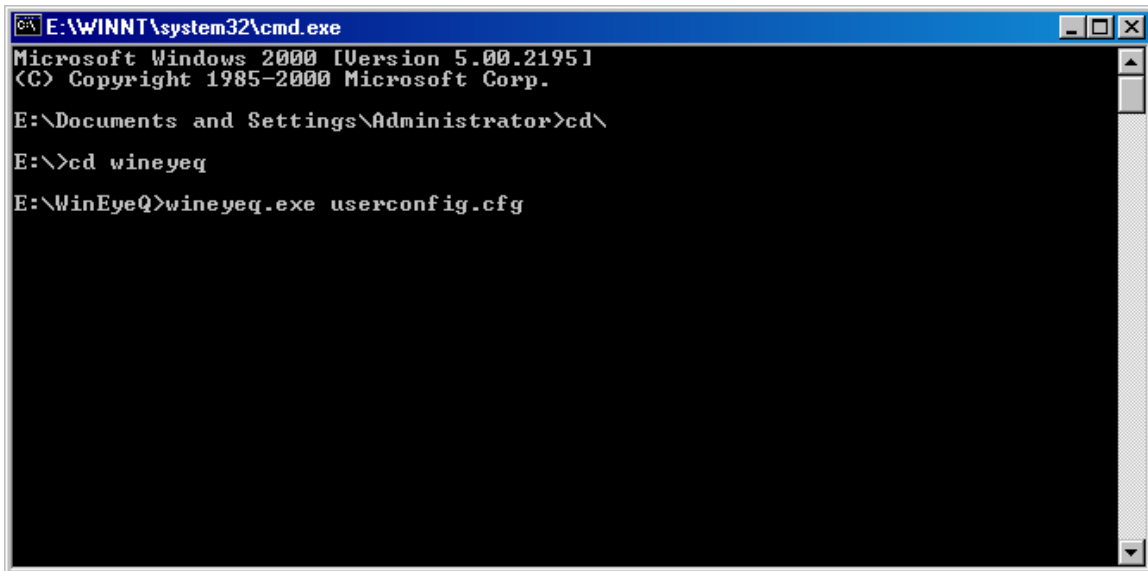


Creating Custom Scenarios

The first step in creating a custom scenario is to save and rename the WinEyeQ.cfg file. Be sure that the new configuration is a .cfg file and not a .txt file. Next, open the parameter files you wish to alter in Notepad, look through the various options available, and alter the parameters as you wish. Be sure to save the files and rename them, again making sure that they are not .txt files. Finally, in your new WinEyeQ configuration file, change the file names so that it points to the appropriate new parameter file(s) you created, and be sure to save the file. Now, you may run your new scenario when you are on the command line in your directory.

Running Custom Scenarios

In order to run a custom scenario that you created, first make sure that all of the files you made and altered are in the WinEyeQ directory. Next, open up the command prompt, again opening up the WinEyeQ directory. To run your file, type in the command "wineyeq.exe [your replacement for the WinEyeQ.cfg file].cfg and press enter.

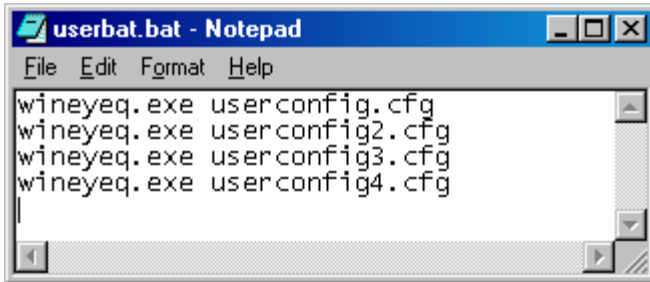
A screenshot of a Windows 2000 command prompt window. The title bar reads "E:\WINNT\system32\cmd.exe". The window content shows the following text: "Microsoft Windows 2000 [Version 5.00.2195] (C) Copyright 1985-2000 Microsoft Corp." followed by a series of commands and their outputs: "E:\Documents and Settings\Administrator>cd\" (output: E:\), "E:\>cd wineyeq" (output: E:\wineyeq), and "E:\WinEyeQ>wineyeq.exe userconfig.cfg". The window has a standard Windows 2000 interface with a blue title bar and a scroll bar on the right.

```
E:\WINNT\system32\cmd.exe
Microsoft Windows 2000 [Version 5.00.2195]
(C) Copyright 1985-2000 Microsoft Corp.

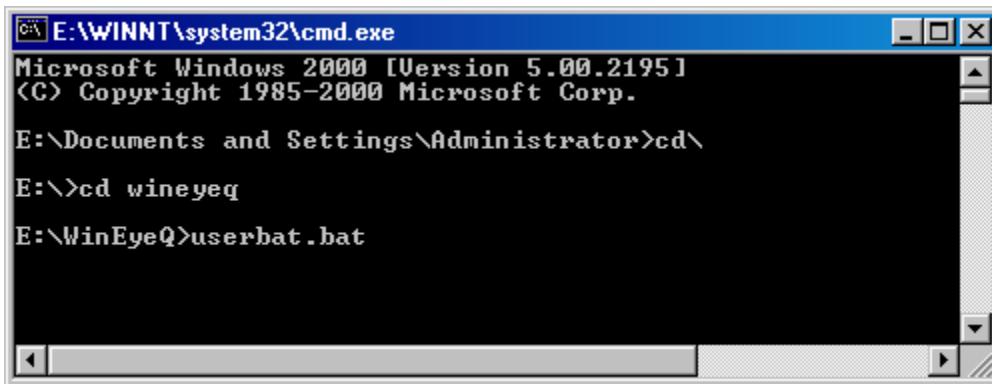
E:\Documents and Settings\Administrator>cd\
E:\>cd wineyeq
E:\WinEyeQ>wineyeq.exe userconfig.cfg
```

Creating and Running .bat files

You can create files that run several custom configurations in sequence by creating a .bat file. First, you must create each individual configuration you wish to run, using the steps mentioned earlier, and be sure that they are all given unique file names within the WinEyeQ directory. Next, open up a Notebook document, and enter the commands you wish to use in sequence, being sure to press enter after each command. Save the file when you are done as a .bat file in the WinEyeQ directory.



Finally, when you are in the command interface, simply enter the .bat file name and press enter to run the sequence.



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Appendix A

Theoretical Maximum MOS Scores and R Factors

The following chart contains the theoretical maximum values for Listening and Conversational MOS and R factor by codec type.

Codec Name	MOS-LQ	MOS-CQ	R-LQ	R-CQ
G.711 u-law	4.2	4.18	93	92
G.711 A-law	4.2	4.18	93	92
G.722 64k	3.88	3.84	94	93
G.722 56k	3.73	3.69	90	89
G.722 48k	3.53	3.48	84	83
G.722.1 32k	4.04	4.01	100	99
G.722.1 24k	3.91	3.91	96	95
G.722.2 23.85k	4.164	4.14	106	105
G.722.2 23.05k	4.16	4.14	106	105
G.722.2 19.85k	4.16	4.14	106	105
G.722.2 18.25k	4.09	4.09	103	102
G.722.2 15.85k	4.09	4.06	102	101
G.722.2 14.25k	4.06	4.04	101	100
G.722.2 12.85k	3.98	3.95	98	97
G.722.2 8.85k	3.73	3.69	90	89
G.722.2 6.6k	3.35	3.3	79	78
G.723.1-5.3k	3.61	3.57	74	73
G.723.1-6.3k	3.77	3.73	78	76
G.726-16k	2.82	2.77	57	56
G.726-24k	3.35	3.3	68	67
G.726-32k	4.04	4.01	86	85
G.726-40k	4.16	4.14	91	90
G.728	4.04	4.01	86	85
G.729/G.729B	3.95	3.91	83	82
G.729A/G.729AB	3.91	3.88	82	81
G.729E 8.0k	3.91	3.88	82	81
G.729E 11.8k	4.11	4.09	89	88
AMR NB 12.2k	4.09	4.06	88	59
AMR NB 10.2k	3.91	3.88	82	81
AMR NB 7.95k	3.69	3.65	76	75
AMR NB 7.4k	3.61	3.57	74	73
AMR NB 6.7k	3.44	3.39	70	69
AMR NB 5.9k	3.25	3.21	66	65
AMR NB 5.15k	3.06	3.02	62	61
AMR NB 4.75k	3.02	2.96	61	60
iLBC 13.3k	3.88	3.84	81	80
iLBC 15.2k	3.95	3.91	83	82
Speex NB 2.15k	2.92	2.87	59	58
Speex NB 5.95k	2.92	2.87	59	58
Speex NB 8k	3.39	3.35	69	68
Speex NB 11k	3.88	3.84	81	77
Speex NB 15k	4.11	4.09	89	88
Speex NB 18.2k	4.11	4.09	89	88
Speex NB 24.6k	4.16	4.14	91	90
Speex NB 3.95k	2.41	2.36	49	48

Appendix B

Sample SNMP traps.

This would be an Alert for an Audio stream that had low packet interval:

```
1.3.6.1.4.1.27631.1 WinEyeQ
1.3.6.1.4.1.27631.1.1 Alert
1.3.6.1.4.1.27631.1.1.1 Audio
1.3.6.1.4.1.27631.1.1.1.3 Interval
1.3.6.1.4.1.27631.1.1.1.3.1 Packet Interval Low Alert
1.3.6.1.4.1.27631.1.1.1.3.2 Source 120.249.50.100:50354
1.3.6.1.4.1.27631.1.1.1.3.3 Dest 120.249.50.75:25008
1.3.6.1.4.1.27631.1.1.1.3.4 Threshold 58.80
1.3.6.1.4.1.27631.1.1.1.3.5 Value 58.42
1.3.6.1.4.1.27631.1.1.1.3.6 Time 05/08/2007 17:43:15.397
1.3.6.1.4.1.27631.1.1.1.3.7 ID = BB94C810992CEE9D58731FB3D16753F2 -
User = 500095
```

And this would be an Alarm for a Video stream that had high jitter:

```
1.3.6.1.4.1.27631.1 WinEyeQ
1.3.6.1.4.1.27631.1.2 Alarm
1.3.6.1.4.1.27631.1.2.2 Video
1.3.6.1.4.1.27631.1.2.2.2 Jitter
1.3.6.1.4.1.27631.1.2.2.2.1 Jitter High Alarm
1.3.6.1.4.1.27631.1.2.2.2.2 Source 120.249.50.100:50354
1.3.6.1.4.1.27631.1.2.2.2.3 Dest 120.249.50.75:25008
1.3.6.1.4.1.27631.1.2.2.2.4 Threshold 1.10
1.3.6.1.4.1.27631.1.2.2.2.5 Value 3.09
1.3.6.1.4.1.27631.1.2.2.2.6 Time 05/08/2007 17:43:16.697
1.3.6.1.4.1.27631.1.2.2.2.7 ID = BB94C810992CEE9D58731FB3D16753F2 -
User = 500095
```


Appendix C

Call Scoring - Letter Grades

The letter grades assigned to the various scores associated with signaling and audio and video quality are determined by the following scale:

A+	98 or above
A	92 or above
A-	90 or above
B+	88 or above
B	82 or above
B-	80 or above
C+	78 or above
C	72 or above
C-	70 or above
D+	68 or above
D	62 or above
D-	60 or above
F	Below 60

Audio Scores

This score is designed to provide a comprehensive value which considers all components that contribute the overall QoS/QoE of an audio stream.

The audio score is calculated using Touchstone's proprietary formula which considers the stream's jitter, inter-packet interval, early arrival packets, late arrival packets, listening and conversational MOS scores, listening and conversational R Factor scores and weights these factors against their optimal values.

The audio scores use the scale identified above to determine a corresponding "grade" for the stream.

Video Scores

This score is designed to provide a comprehensive value which considers all components that contribute the overall QoS/QoE of a video stream.

The video score is calculated using Touchstone's proprietary formula which considers the stream's jitter, picture rate, early and late packets to assess the overall stream quality. This score is graded using the scale outline above.

Media Score

This score is designed to provide a comprehensive value which considers all components that contribute the overall QoS/QoE of stream.

The media score is calculated using Touchstone's proprietary formula which considers the weighted values obtained from the scoring of the audio and video components.

The media score uses the scale identified above to determine a corresponding "grade" for the media component.

Signaling Score

This score is designed to provide a comprehensive value which considers all components that contribute the overall QoE of the signaling component of a SIP or H.323-based telephony call or video conference.

The signaling score is calculated using Touchstone's proprietary formula which considers signaling metrics such as initial response time, post-dial delay and call teardown time. These metrics are then compared against industry-accepted values and the score is then calculated based upon the results.

The signaling score uses the scale identified above to determine a corresponding "grade" for the media component.

Overall Score

The overall score is designed to provide a comprehensive value which considers all components that contribute the overall QoE of a SIP or H.323-based telephony call or video conference.

The overall score is calculated using Touchstone's proprietary formula which considers a weighted combination of the signaling and media scores. This score paints a highly accurate portrait of the overall quality of the session.

The overall score uses the scale identified above to determine a corresponding "grade" for the call or conference.

WinEyeQ User's Guide

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