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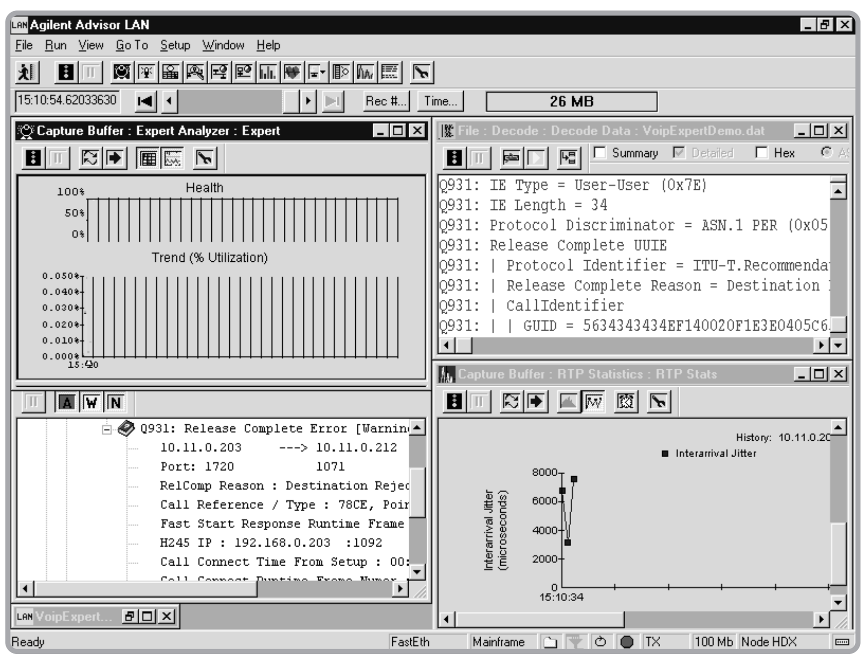
Agilent Advisor with Voice Personality – J4618B

Product Overview

The Agilent Advisor with Voice Personality is a Windows® based software solution that adds Voice over IP analysis to the Advisor and creates the personality known as the Advisor with Voice. The Advisor with Voice offers real-time troubleshooting for Voice over IP (VoIP) applications on LAN, WAN, and ATM networks. Expert analysis tracks each state of the VoIP call control and voice transport process and automatically detects errors and protocol anomalies.

Packets using RTP to transport voice or video are analyzed for errors. Sessions and participants are identified, and packet loss and jitter are measured for each stream or conversation. In addition, detailed examination, packet by packet, is offered through clear and accurate decodes and real-time filtering. The Advisor with Voice analyzes the most extensive list of VoIP protocols in the industry today, augmented now with the addition of MEGACO/H.248 and H.225.0 Annex G.

RTP measurements, VoIP Expert analysis, packet decodes, and statistics can be run simultaneously using Advisor with Voice. These analysis features allow users of Voice and Fax over IP networks and services to evaluate equipment such as gateways and gatekeepers and troubleshoot and optimize IP Telephony deployments. Advisor with Voice assists in identifying internetworking, interoperability, and performance problems. It is especially useful during multi-vendor network integration and performance assessment.



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Innovating the HP Way

Key Features

- Packet loss and jitter calculated for each individual RTP stream
- Graphs the distribution of RTP packet loss and jitter values and over time
- Comprehensive, real-time Expert Commentators for VoIP Signaling and Voice packet transport.
- Replay the analysis for previously captured voice packet sessions.
- Export information on all RTP packets to a spreadsheet.
- Automated hierarchical process for troubleshooting and traffic optimization
- Call Detail Records for each call or part call
- Real-time decodes, filtering and analysis on LAN, WAN and ATM
- The most Extensive list of VoIP decodes available including recent drafts
- Simultaneous RTP and Expert analysis with decodes and statistics

The Challenges for IP Telephony

Voice and video communications are inherently real-time applications. In order to achieve conversation over telecommunication networks, you must signal between the voice and video terminals to ensure both ends are active, responsive, and can agree on a common set of capabilities to communicate and exchange features. IP Telephony is a technology in its infancy and the industry faces at least two significant challenges. First, to ensure virtual connection across this connectionless packet IP network is achieved using new protocol standards, such as H.323; MGCP; MEGACO/H.248; or SIP. Second, is to transport packets over the IP network in a timely manner with high integrity, thereby ensuring acceptable voice and video quality. Engineers skilled in Voice and Fax over IP technology are working hard to overcome these problems. Challenges exist because of the following reasons:

- New signaling protocols such as MEGACO, SIP, and MGCP are far from complete, contain optional functionality, and are being revised at a fast rate. Interoperability problems between equipment suppliers often occur, especially as vendors adhere to different releases of the same standard or implement options not supported by the other party. Devices may not reset gracefully when unexpected behavior is encountered.

- Standards leave room for interpretation leading to connectivity or feature negotiation problems among different vendor systems.

- Voice traffic requires a minimum quality of service from the packet-switched network to allow acceptable speech quality. Error rate, packet loss, jitter, and delay must be measured and IP networks must be upgraded with mechanisms to ensure acceptable QoS.

IP Telephony Solutions on the Advisor

The Advisor with Voice (J4618B) is a software solution designed to run exclusively on the Advisor test platform. The Advisor with Voice provides a comprehensive set of features to diagnose problems and performance bottlenecks in VoIP networks. Solutions for both IP QoS and signaling are offered. Packet loss and jitter for each RTP session is measured in real-time. The time of arrival of each packet is recorded and a comparison made with the time stamp and sequence number contained in the RTP header. Using this method, an accurate measurement of jitter and packet loss introduced up to that point in the network can be made. Results are tabulated and graphed, then made available for export to external analysis tools, such as spreadsheets. Voice packets can be captured on WAN, LAN, and ATM interfaces, and replayed through the application to reexamine the analysis.

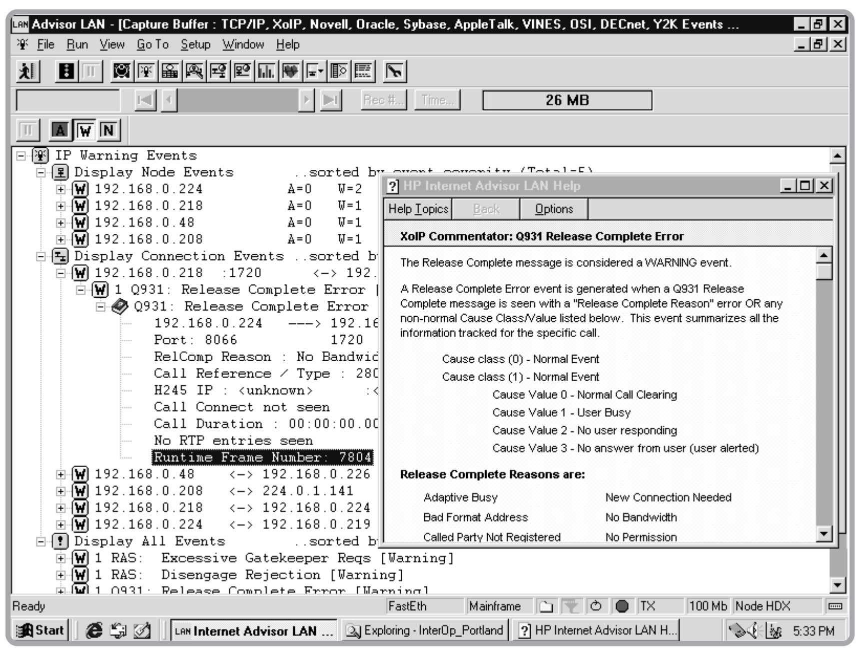
This software package also contains the VoIP Expert Commentators for H.323. The VoIP Expert Commentators intelligently analyzes every stage of the call process and informs the user of all errors and notable events. Further, the Advisor with Voice's features deliver clear, accurate, color-highlighted decodes and real-time filters to select only those signaling or voice packets of interest. The most extensive list of Voice and Fax over IP protocols in the industry is supported.

The Value in Using the Advisor with Voice and Expert Commentators in the Test Laboratory

In the test laboratory, you might need to evaluate the quality of the voice delivered by a gateway. This evaluation is usually determined by two factors. The DSP (Digital Signal Processing) chips inside the gateway share their processing power between calls. When the number of simultaneous calls approaches the capacity of the gateway, codecs will not possess sufficient resources to construct voice packet correctly. Alternatively, the gateway must be evaluated for its ability to preserve voice quality when IP packets are dropped, arrive in the wrong sequence, are delayed, jittered or corrupted. The Advisor with Voice is the ideal solution to measure and record accurate statistics for RTP voice packet performance. VoIP devices do not all implement silence suppression in the same way, where standards may not be explicit in how this should be done. The Advisor with Voice tracks each voice session and troubleshoots those dropped due to silence suppression.

Analysis for each session can be exported to spreadsheets for customized packet by packet analysis. For example, components within gateways such as output queues and dejitter buffers can be configured and optimized. Peaks in jitter can be located and correlated with other events claiming excessive network resources.

The VoIP Expert Commentators save many hours in the test lab because problems with the functioning of the system under test are highlighted automatically. Specialists designing VoIP products can concentrate on building competitive advantages into their products and can leave the VoIP Advisor with Voice to ensure the rules of the H.323 standard are obeyed by their design. Compliance with the standard is important because when that product is connected to other vendor's products, compliance to the standard will ensure interoperability. A deviation from the standard in the behavior of vendor A's H.323 stack may not cause a problem when they connect to themselves. However vendor B's product may react in an unpredictable way to this quirk and tear down the call. The analyzer detects this "non-compliant" behavior during real-time monitoring and the user can devote attention to solving other problems.

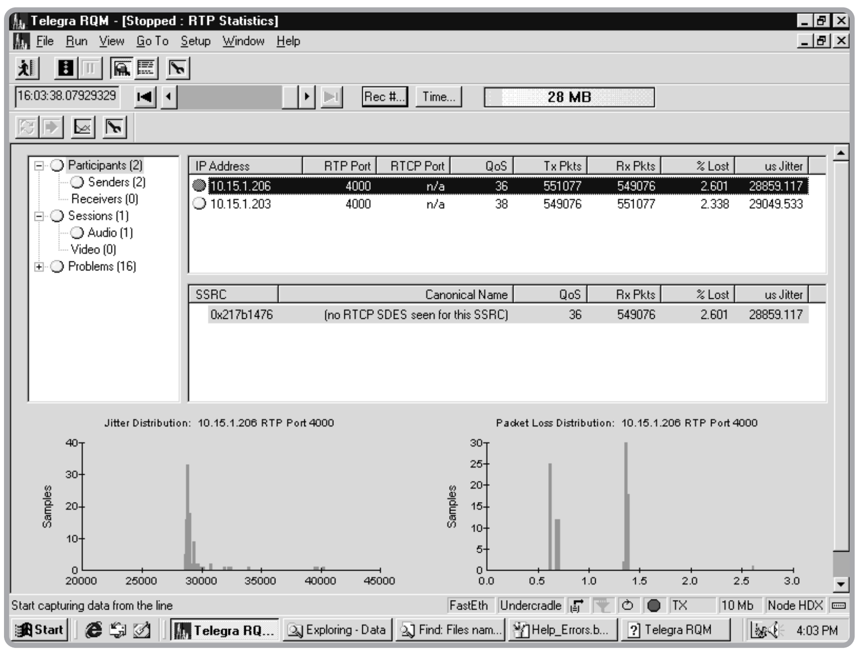


The VoIP Expert Commentators are also useful for code optimization. For example a VoIP terminal receiving Call Clear sends a Call Clear back to the "Clearing" terminal. This process is noncompliant behavior. And is not necessarily catastrophic. However, it is inefficient and may cause problems with other vendors' equipment.

The Agilent VoIP Expert Commentators watch for the complete spectrum of problems simultaneously - including the unexpected. It would be impossible for the human brain to watch for hundreds of problems at the same time in this way. The user is alerted to all problems and left to make the decision whether that problem is significant for the current task being engineered. For example, slow gatekeeper response time is notified. The user decides if this is significant for the current test objective.

The Value in Using the Advisor with Voice and Expert Commentators in Operational VoIP Networks

The Advisor with Voice is used to measure the level of impairments introduced by an IP network so that an assessment can be made whether acceptable speech quality can be expected over that network or whether poor speech quality is due to the IP network performance. In an operational network, it is vital to isolate the cause of poor voice or video transport. Such poor quality may be due to packet impairments in the IP routers or over utilization of VoIP gateway. Measurement of packet loss, packet loss ratios, and jitter are an essential part of isolating the cause of poor voice quality in the IP network.



The Advisor with Voice data is also available for incorporation into reports on the performance of the network using external software packages.

Call Control of voice or video conferencing over IP contains complexity at every stage. The VoIP Expert Commentators on the Advisor with Voice offers the most sophisticated technology available today for automated troubleshooting of VoIP signaling and voice packet transport over operational IP networks. The following is a selection of typical problems detected by this tool:

- Gatekeepers, Location Services Directories or SIP Redirect Servers slow to respond, out of service, slow to update new user information or not knowing the address of the called party

- Call setups experiencing problems due to congestion or router failure preventing signaling packets reaching the destination; the called party being out of service, slow to answer or unable to redirect calls when busy

- Terminal equipment incapable of supporting the common set of features and services required or insufficient bandwidth to hold the intended conference

- Terminals fail to back off gracefully in the event of failed negotiation. This process may result in network resources being unusable

- Voice quality will suffer due to insufficient bandwidth or high latency across the IP network. Voice quality may degrade when calls may terminate due to congestion arising along the path of the call or within the terminal equipment itself where multi-tasking may divert resources to a data application.

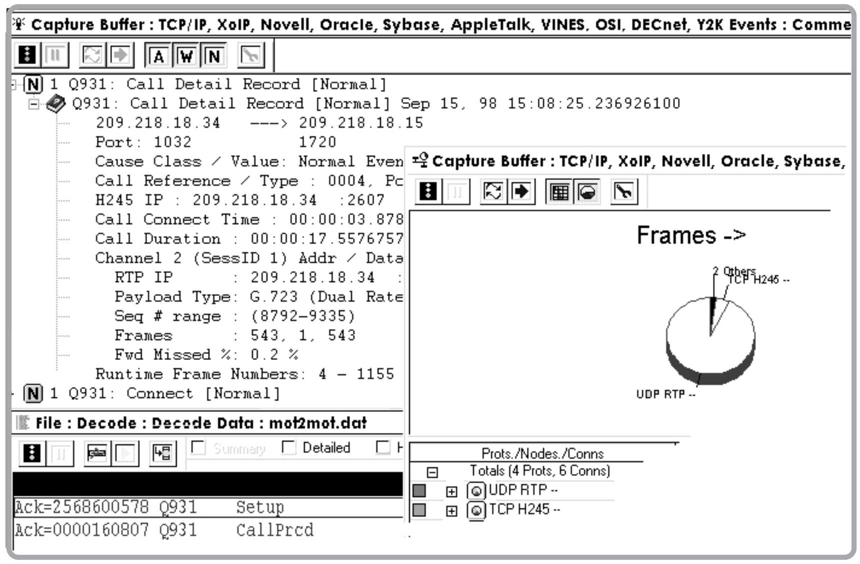
The H.323 protocol is new to many, but is also very complex. Engineers new to the intricacies of the protocol can troubleshoot new VoIP installations. With every stage of this process, the VoIP Expert Commentators on the Advisor with voice give comprehensive and automated analysis.

Automated Hierarchical Process to Troubleshooting

The VoIP Expert Commentators detect and inform the user of errors and anomalies. Expert HELP text is available to explain why the error may be a mission-critical problem. Should the user wish to view a detailed decode, one mouse “click” will display the culprit packet or packet ranges. Similarly, the VoIP Expert Commentators may return an “Insufficient Bandwidth” warning. One mouse “click” from the Expert display will show statistics for each node, connection and protocol. The bandwidth being used by both data and voice applications is easily seen.

Once in the decode display, the Advisor with Voice will give a clear and accurate decode of each field in every packet in real-time. Each protocol layer will be displayed in its own color and a line of English text is given for each field or parameter. All layers of the protocol stack, including Ethernet, IP, TCP/UDP, and the Voice and Fax protocols, are fully reassembled and decoded.

Filtering capabilities allow the user to select only those packets of interest. For example, perhaps only the SIP/SDP signaling is under suspicion. When problems are being reported only with calls to a certain IP address, the Advisor with voice can be set up to capture only the signaling to that destination. There may be 100,000 packets per second on a Fast Ethernet link carrying voice. However the Advisor with Voice’s real time filtering abilities will display only the packets of interest and a quick glance at the display will show the user if any calls have been made to the defective destination.



A distinct advantage of the Advisor with Voice is its ability to analyze in realtime as well as in Post-Analysis Mode. The Post-Analysis method involves attaching the analyzer to the network, capturing data for a period of time, and then stopping the analyzer. The analysis is then performed OFF LINE, while no more data is taken from the line. This is a time consuming process, especially when the problem packet is not found in the capture buffer.

Real-time mode, as implemented by the Advisor with Voice allows problem analysis while connected to the wire and capturing data. This process is much faster, problems can be found quicker, and customers' network can be restored to health in much shorter time. Furthermore, the analyzer can be left overnight and relied upon to catch the problem packet. The user comes in the next morning knowing the fault is captured!

Agilent Advisor with Voice Personality Capabilities

Decodes for All VoIP Protocols

The Advisor with Voice delivers clear, accurate color highlighted decodes and real-time filters to pick out only those signaling or voice packets of interest. The most extensive list of Voice and Fax over IP protocols in the industry is supported. These include:

- H.323 version 3; comprising: H.245, H.225.0 (Q.931 and RAS)
- H.261 and H.263 for video conferencing
- IETF SIP, SAP and SDP
- SGCP from Bellcore
- MGCP from the IETF for device control in Carrier-Class IP Telephony
- NCS (PacketCable's™ profile of MGCP) for IP Telephony over the CableTV plant
- T.38 for real-time Fax over IP
- RTP and RTCP for transport of voice packets over IP
- MEGACO/H.248. jointly agreed by IETF and ITU-T for Carrier-Class IP Telephony
- H.323 v3; Annex E & F and H.225.0 Annex G
- SCTP from IETF SIGTRANS Working Group

This analysis can be performed in realtime or postprocess, capturing data from LAN, WAN, or ATM interfaces. The Advisor with Voice has a track record for being first to market with new VoIP protocols, so your investment is protected and you can depend on your test equipment to keep pace with your VoIP innovations.

This functionality is standard to the Advisor with Voice, with the installation of Release 11.4 system software.

Filters

- Capture filters select only the defined packets for storage to the capture buffer. Packets can be filtered into the buffer for the following VoIP protocols easily, using the GUI:

- H.225.0 RAS	- SGCP
- H.225.0 Call Signaling (Q.931/Q.932)	- RTP
	- RTCP
- MGCP	- SIP

- Packets in the capture buffer can be searched for by protocol. The cursor is placed over the packet once it is found.

- Once captured in the buffer, further filtering can select those packets to be displayed on the screen. This process is achieved using the display filters. Display Filters are available for the following VoIP protocols:

- H.225.0 RAS	- MGCP
- H.225.0 Call Signaling (Q.931/Q.932)	- SGCP
	- RTP
- H.245	- RTCP
- H.450	- SIP
	- SDP

- These filters are currently functional for analysis using Ethernet 10/100Mbps; FDDI and Token Ring interfaces:

This functionality is standard to the Advisor with Voice, with the installation of Release 11.4 system software.

VoIP Expert Commentators

Automatically detects Errored Call Set-Up and Tear Down. For example:
- Unreachable Destination
- User busy
- Resources unavailable
- Interworking error

Generates Alarms for non-Standard Protocol Behavior. For example:
- Invalid message
- Unknown data type

Warns of Errors. For example:
- Open Logical Channel Reject
- No bandwidth
- Resource unavailable
- Security denial
- Transport QoS not available

Measures Gatekeeper Performance. For example:
- Alarms on excessive requests
- Alarms on long response times

Alerts the user to slow IP network and VoIP device performance. For example:
- Long call set up times
- Missed sequenced and duplicate RTP packets

Draws attention to VoIP device incompatibility. For example:
- Terminal capability set reject or release

VoIP Expert Commentators Performance Measurements

A comprehensive set of performance measures is made automatically by the VoIP Expert Commentators. These include:

- Call setup time Measurement
- Measurement of the response time of gatekeepers and gateways
- RTP packet performance, logging duplicate and lost packets by sequence number

Manual measurements can be accomplished using the delta timestamp mode of the summary decode.

This functionality is added to the Advisor with Voice, with the installation of the J4618B software.

Call Detail Records

The VoIP Expert Commentators produce a record of all H.323 calls monitored. Comprehensive information is logged regarding each call including:

- Call Set up Time
- Call Duration
- Call Clear Cause
- Terminal Capabilities Negotiation
- Payload Type
- Number packets sent
- IP and UDP addresses

This functionality is added to the Advisor with Voice, with the installation of the J4618B software.

Agilent Advisor with Voice Personality

The Agilent Advisor with Voice Personality software application measures the IP packet performance of Voice over IP networks. Specifically, packet loss and jitter is measured for each session or conversation using RTP to transport voice or video.

Standard Agilent Advisor Capabilities

The following features are standard with all Advisors

Packet, Frame and Cell Capture

All data can be captured in up to 64 Mbytes of buffer via all LAN, WAN, and ATM interfaces at full line rate. Each packet is stamped with a 100 ns resolution timestamp representing the start of the packet.

Real-time and Post-Process Analysis

As they are captured, packets are displayed during run-time. In addition, protocol packets can be displayed from the capture buffer after they have been captured (this is known as post-process analysis). Expert analysis is also offered in run-time.

Connection, Node and Protocols Statistics

Counts for traffic levels for each Connection pair (IP address pair) and each node. All protocols seen are retained. Numbers of frames, bytes, and utilization levels are displayed for each category. The user can elect to view these counts in any order. In other words, if the user wants to view protocol distribution for all addresses, “PROTOCOL” must be selected first. The user can then drill down to see usage for each connection or node within each protocol.

This display can be used to measure the average bandwidth used for voice, voice signaling, and data applications as defined by the protocol. Similarly, if “CONNECTIONS” is selected for top-level viewing, the user can drill down to see the traffic levels for each application, by protocol, between the two workstations making up the connection.

Presentation

Decoded packets are displayed in summary, detailed, and hexadecimal formats.

- The Summary window shows timestamp, source and destination address, protocol type, and further information, such as RTP-Payload Type
- The Detailed format breaks out all header fields and displays each field with its own color
- The Hexadecimal window displays all bytes for each frame in hexadecimal, retaining the color-coding used in the "Detailed" display. The payload is also displayed in hexadecimal
- All data can be printed or stored to a file for later analysis. Search and display filter capabilities are supported for IP, TCP, and UDP data
- The "Hexadecimal to detail" mapping feature allows the user to cursor to a field in the Detailed window and the same bits are selected in the Hexadecimal window
- Display options allow all protocol fields to be viewed in the summary window or alternatively only the IP addresses plus the top protocol layer. For example: H.225, H.245, or SIP and SDP
- Filters can be configured quickly with a right mouse click on the packet of interest and selection of a Mac layer or IP layer source and destination address

Configuration Information

To achieve full functionality as described in this document, the J4618B RQM Software with VoIP Expert Commentator's release 11.4 requires an Advisor with system software release 11.4 or later. For real-time analysis, this software solution requires any LAN measurement interface. The Advisor platform must be a C or D model, such as J3446C/D, J2300C/D, or J3754C/D. This software offers post-analysis functionality when used with WAN or ATM Advisor measurement interfaces.

Technical Specifications

This section defines the specification to which the decode solutions comply.

- ITU-T recommendation T.38, Procedures for real-time Group 3 facsimile communication over IP networks (June, 1998)
- “Media Gateway Control Protocol (MGCP) Version 0.1” IETF Draft <draft-huitema-megaco-mgcp-v0r1-05.txt>, Internet Engineering Task Force, (February 21, 1999)
- “Simple Gateway Control Protocol (SGCP) Version 1.1 Draft”, IEFT Draft <draft-huitema-sgcp-v1-02.txt>, Internet Engineering Task Force, (July 30, 1998)
- RFC 2327, “SDP: session description protocol”, (April 1998)
- RFC 1889, RTP: A Transport Protocol for real-time Applications (January 1996)
- RFC 1890, RTP: Profile for Audio and Video Conferences with Minimal Control (January 1996)
- RFC 2543, SIP: Session Initiation Protocol (March 1999)
- ITU-T, Recommendation H.323, version 3 “Visual Telephone Systems and Equipment for Local Area Networks which Provide Non-Guaranteed Quality of Service”
- ITU-T, Recommendation H.225, version 3 “Call Signaling Protocols and Media Stream Packetization for Packet Based Multimedia Communications Systems”
- ITU-T, Recommendation H.245, version 5, “Line Transmission of Non-Telephone Signals”
- RFC 2032 referring to Video codec for audiovisual services at p x 64 kbit/s ITU-T (International Telecommunication Union - Telecommunication Standardization Sector) Recommendation H.261, (1993)
- PacketCable Networked-based Call Signaling protocol specification PKT-SP-EC-MGCP-101-990312 (March 12, 1999)
- MEGACO/H.248 Internet Engineering Task Force, (February 05, 1999) <draft-ietf-megaco-protocol-05.txt>

Related Literature

Agilent Advisor LAN	Product Overview	5980-0990E
Agilent Advisor WAN	Product Overview	5967-5566E
Agilent Advisor ATM/WAN	Technical Specifications	5980-0786E
Agilent Advisor ATM	Product Overview	5968-1437E
Internet Reporter	Technical Specifications	5968-6188E
Agilent Advisor	Brochure	5980-1093E
Agilent Advisor An Introduction to Sessions Initiation Protocol	White Paper	5968-6297E
Agilent on Troubleshooting H.323 Signaling	Whiter Paper	5968-3642E
Agilent Advisor Troubleshooting H.323 Signaling	App Note	5968-4450E

Warranty

Hardware: 1 year
Software: 90-day replacement only

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Job Title ______________________

Business Mailing Address ______________________

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City ______________________

State/Province ______________________

Country ______________________

Zip Code ______________________

E-Mail Address ______________________

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Phone Number ______________________
(incl. area/country code)

Fax Number ______________________

Do you have a budget set for this application?

- ☐ Yes
- ☐ No
- ☐ In process

What is your time frame to implement this product?

- ☐ 30 days
- ☐ 180 days
- ☐ 90 days
- ☐ Other – (please define) ______________________

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Product(s) of Interest

☐ ***The Agilent Advisor*** – Integrated, High-Performance Troubleshooting for:

____ Advisor LAN
____ Advisor WAN
____ Advisor ATM

☐ ***The LAN Analyzer*** – Scaleable Ethernet and Token Ring Test Solutions

☐ ***Telegra Fax Test*** – Fax Protocol and Low Generation Analysis

☐ ***Telegra Voice Quality Tester*** – Detailed Voice Analysis for Clarity, Echo and Delay using PSQM and PAMS

☐ ***Telegra Voice and Fax over IP*** – Protocol Analysis

☐ ***FASTest*** – Automated Service Verification for PSTN and IP Networks

What is the main problem you need to solve on your network?

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Notes

Notes

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This Product is Y2K Compliant

Agilent Ordering Information

J4618B	Agilent Advisor with Voice Personality

Platforms

J2300D	Agilent Advisor WAN
J3446D	Agilent Advisor LAN — Fast Ethernet
J3447A	Agilent Advisor LAN — Fiber Interface for J3446C
J3444A	Agilent Advisor LAN — Fast Ethernet undercradle

Instructor Led Training CBT

H7211A/B	Essentials of VoIP Protocols
Opt 207	Instructor Led Training

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