

# JDSU J6844A Telephony Network Analyzer

## Technical Overview



### The Most Powerful VoIP Troubleshooting Tool Available

The JDSU Telephony Network Analyzer simplifies and expedites the resolution of quality available and signaling problems in IP telephony networks. It provides simple and precise diagnostics of VoIP Quality of Service (QoS) through non-intrusive measurements, including new voice quality measurement technology known as Predictive Mean Option Score (MOS). It also provides simplified troubleshooting of call signaling and control through embedded expert analysis of VoIP protocols. As an JDSU Network Analyzer solution, the Telephony Network Analyzer includes complete layer 1-7 testing over LAN, WAN, and ATM networks. This powerful tool offers the most advanced VoIP signaling and service quality troubleshooting capabilities available:

- Non-intrusive voice quality measurements using breakthrough Predictive MOS technology
- Simple analysis that exposes the impairments to voice quality
- Precise measurements of IP network performance for VoIP services
- Diagnostics for troubleshooting and identifying problems at the root cause
- Simplified troubleshooting of call signaling and control to quickly turn-up or restore service
- Voice quality and signaling diagnostics for MPLS and IPv6 networks

### The Telephony Network Analyzer Meets the Challenges of Deploying VoIP

Integrating commercial or enterprise voice services on a data network can offer significant capital and operational cost reductions, and can enable new competitive services. However, deploying VoIP is a difficult and complicated series of tasks, and presents numerous challenges that can delay or even prohibit the delivery of services. These challenges include:

- **Implementing complex VoIP call signaling**

There are many different VoIP signaling protocols in use today. Each one is undergoing constant development, and offers many options and variances. This results in many interoperability problems between systems and equipment, especially as vendors adhere to different releases of the same standard or implementation options not supported by another vendor.

- **Accelerated service turn-up and restoration**

Competitive pressure, reductions in operational spending, and greater demand for new services necessitate faster service turn-up and resolution to problems. New technologies for VoIP increase the complexity of services, making it more difficult for technicians and engineers to accomplish rapid problem resolution.

- **Meet service quality expectations**

To win and retain customers, voice service quality must meet customer expectations set by traditional circuit-switched networks. VoIP requires a minimum quality of service (QoS) from the data network to meet service objectives. VoIP packet loss, jitter, and delay must be tightly controlled with complex network QoS mechanisms.

- **Troubleshooting complex multi-service networks**

When calls don't complete or customers complain about service quality, how can you quickly determine the problem so you can fix it? Identifying that a problem exists is only the first step one must then be able to troubleshoot and fix problem if service is to be restored and customers retained.

The JDSU Telephony Network Analyzer significantly simplifies and accelerates the tasks needed in deploying VoIP by addressing these challenges head-on. The Telephony Network Analyzer accelerates the turn-up of new VoIP services and the resolution of customer problems with VoIP services. It offers simple but powerful troubleshooting solutions for VoIP.

The Telephony Network Analyzer enables the design, deployment, and maintenance of VoIP services by exposing signaling, interoperability, and service performance problems at their source. It is a software solution on the JDSU Network Analyzer, and supports LAN and WAN technologies, including Ethernet, ATM, and Frame Relay.

### **Exclusive New Technology for Non-intrusive Voice Quality Scoring**

The JDSU Telephony Network Analyzer offers breakthrough technology for non-intrusive voice quality scoring. Predictive Mean Opinion Scores (MOS) rates the overall voice quality of a VoIP call on the widely recognized MOS scale of 1-5. MOS scores are based on the TNA's non-intrusive analysis of a VoIP call, and are calibrated to match the industry standard for voice quality, ITU recommendation P.862, called the Perceptual Evaluation of Speech Quality (PESQ). Actual PESQ measurements may be obtained with the JDSU Voice Quality Tester (VQT), while accurate predictions of PESQ measurements, reported as MOS, are provided by the Telephony Network Analyzer.

Predictive MOS calculations are based on the known characteristics of specific VoIP gateways and phones that impact voice quality. IP network performance measurements are only half of the voice quality equation. There are many processes within a VoIP gateway, many of them proprietary, that impact voice quality. Different gateways will deliver different levels of quality for a specific level of network performance. In other words, there is no such thing as a "generic gateway", and the TNA addresses this fact by calculating MOS based on actual VoIP network performance measurements matched to accurate calibrations of real VoIP gateways.

The Telephony Network Analyzer is calibrated to specific VoIP gateways. By simply matching an RTP stream's destination IP address to a specific VoIP gateway model, a MOS score is calculated based on how that VoIP gateway model delivers voice quality under the measured conditions of the network.

The Telephony Network Analyzer also calculates "R Factor" scores, based on the ITU recommendation G.107 "E Model". In fact, the TNA can be used to validate actual network performance against network plans per the E Model.

### **Testing and troubleshooting VoIP service quality**

Delivering service quality is considered the main obstacle to winning and retaining customers with VoIP services. The Telephony Network Analyzer (TNA) enables network operators to deliver the service quality their customers expect. It provides complete testing and troubleshooting of VoIP service quality with automatic problem detection that is directly coupled with detailed analysis. With the TNA, engineers and technicians can not only identify a problem, but can then locate the problem and identify its cause to expedite a quick resolution.



***Rapid progression from problem detection to resolution***

The Telephony Network Analyzer provides a unique three-level testing methodology that simplifies the troubleshooting process and enables rapid progression from problem detection to resolution.

**1. Detect problems and identify problem calls**

The TNA's RTCP Monitor provides numerous call statistics and performance data on VoIP calls for high-level monitoring. Monitor the performance of hundreds of calls, in terms of any service parameter, including: MOS, R Factor, packet loss, packet jitter, packet delay, user-defined QoS, and more. Red, Yellow, and Green LED indicators based on performance levels are presented for easy identification of problem calls.

With the RTCP Monitor, you can sort calls based on parameters such as MOS, packet loss, packet jitter, source or destination IP address. You can also filter calls by source or destination IP address. Call statistics and performance data are based on RTCP reports (containing measurements performed by the VoIP endpoints) and on actual RTP measurements performed by the TNA.

By monitoring MOS for calls, the overall end-user quality of each call can be observed, and calls with the worst end-user quality can be filtered for further analysis.

For example, monitor performance of VoIP calls on a network link. Sort by highest packet loss. Select calls with highest packet loss for analysis on the next level: identify and locate problem.

**2. Locate and isolate problems**

Once problem calls have been identified, the TNA can be used to quickly pinpoint the problem. The RTCP Monitor provides automatic drill-down to actual RTP packet measurements for selected calls, along with correlated measurements at three different points in the network for each call. Packet loss, jitter, and delay measurements performed at each of the two VoIP endpoints are reported via RTCP reports. Packet loss and jitter measurements, performed by the TNA at the intermediate point at which the TNA is connected, are reported and correlated with RTCP data.

These capabilities provide indication of the problem (e.g., excessive packet loss or jitter) and where in the network it is occurring. In addition, performance measurements are graphed over time to provide more visibility into the nature of the problem (e.g., bursty loss vs. random loss), and exactly when it occurred.

Performance trending graphs then provide automatic drill-down to a packet-by-packet analysis of key performance measurements. See the precise time and packet at which an impairment, or series of impairments, occurred. Then drill-down to the next level of analysis: identify and troubleshoot the root cause.

**3. Identify and troubleshoot root cause**

From the precise time or packet at which an impairment occurred, you can automatically drill-down to the RTP packet decodes for that call. Precise and simple-to-read decodes for all RTP packets for a call or a stream help determine the source of an impairment.



**VoIP Call Stats**

The Telephony Network Analyzer reports many VoIP call statistics and performance measurements, including:

**Quality Metrics:**

Predictive Mean Opinion Score (MOS):  
current sample  
average per stream  
minimum per stream

R Factor  
current sample  
average per stream  
minimum per stream

QoS (user-defined setting of packet loss and jitter)  
current sample  
average per stream  
minimum per stream

**Diagnostics**

Source and Destination IP address and UDP ports  
Phone number or Synchronization Source (SSRC)  
Call duration  
Call active/inactive indicator  
Number Lost Packets (current sample)  
Percentage Lost Packets (current sample)  
Maximum Percentage Lost Packets since call began  
Average Percentage Lost Packets since call began  
Last RTP jitter  
Maximum Jitter since call began  
Average Jitter since call began  
Jitter Standard Deviation  
Canonical Name of endpoint  
Payload type  
RTP packet count (total and per sample)  
Octet counts (total and per sample)  
Round-trip RTCP packet delay  
Errors  
Alarms

**Real-time Audio Playback**

The Telephony Network Analyzer decodes in real-time the audio in a selected RTP stream, and plays back the audio for subjective listening tests. The audio decode supports G.711, G.729, and G.723.1 codecs. The TNA can also save the decoded audio to a wave file for further analysis. For example, you can use the TNA in conjunction with the JDSU Voice Quality Tester (VQT). Measure end-to-end voice quality with the VQT. To locate where in the network voice quality is impacted, intercept the call with the TNA at any point in the IP network. Capture and save the audio at that point, and measure the voice quality of the captured audio on the VQT.

**Thresholds and Alarms**

The Telephony Network Analyzer allows thresholds to be set for performance measurements, and generates alarms when thresholds are broken. All alarms, along with other events, are automatically logged, so that the Telephony Network Analyzer can be used for unattended testing.

**Phone Number Tracer**

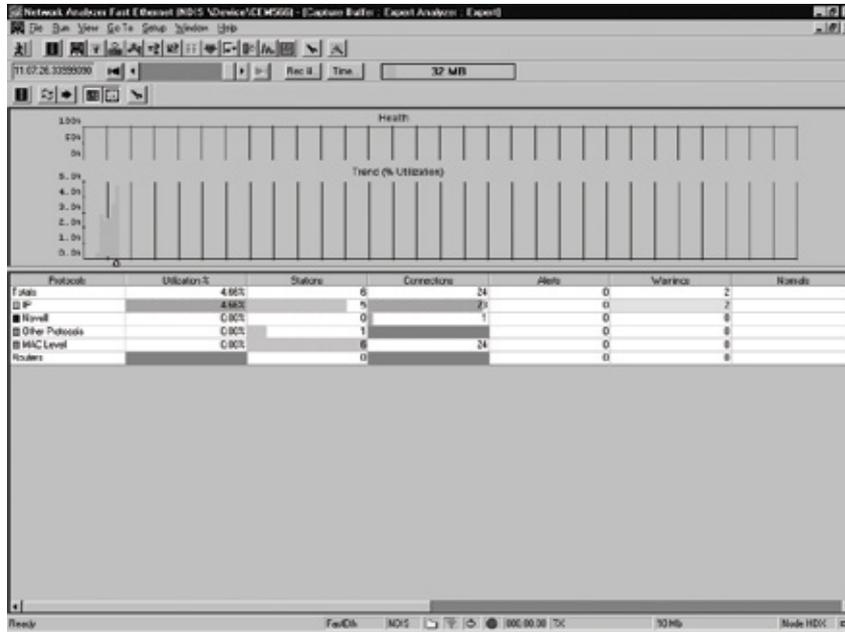
The TNA automatically identifies the calling and called phone number for a call (based on SIP or MGCP signaling) and displays the phone numbers with each RTP stream. This allows the user to quickly find the RTP stream for a user-specific call and to begin troubleshooting. You can also enter the dialed phone number of the call you want to analyze, and based on SIP or MGCP signaling, the Telephony Network Analyzer will automatically find the RTP stream for that call and display call stats and performance measurements.

**Testing and troubleshooting VoIP signaling**

There are many different VoIP signaling protocols in use today. Each one is undergoing constant development, and offers many options and variances. A simple telephone call requires many different message exchanges anywhere from five to over twenty using one of these protocols. A call with multiple parties or enhanced services can require over a hundred message exchanges! When calls don't complete, how can you quickly determine the problem so you can fix it?

The Telephony Network Analyzer accelerates the turn-up of new VoIP services and the resolution of customer problems with VoIP services. It offers simple but powerful troubleshooting solutions for VoIP signaling. It provides real-time and post-process analysis of VoIP signaling with Expert Commentators, call detail records, connection stats, accurate protocol decodes, and more.

H.323, MGCP and SIP Expert Commentators automatically detect and highlight anomalies in VoIP signaling.



TNA Expert Commentator

**Automatically detects errors in 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

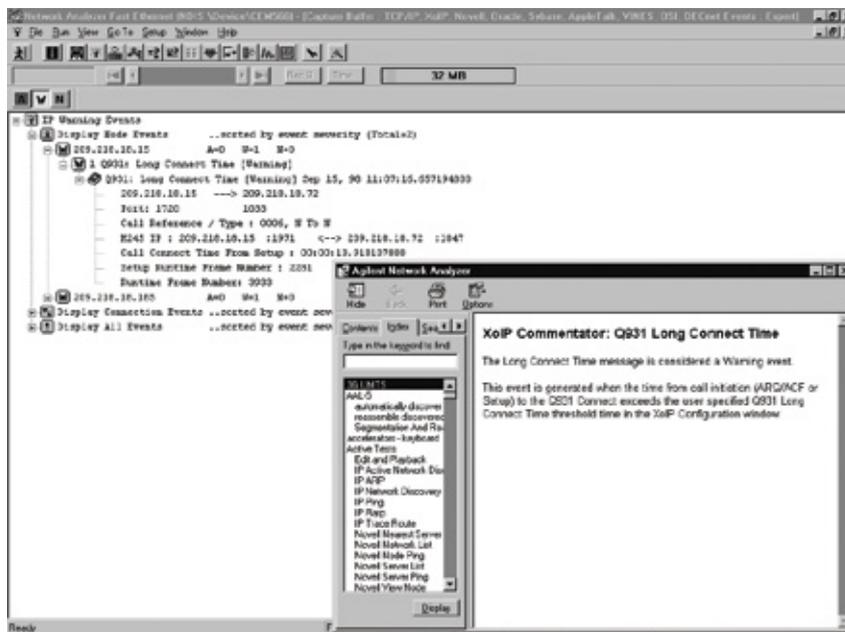
**Draws attention to VoIP device incompatibility. For example:**

- Terminal capability set reject or release

**Performance Measurements**

**In addition, a comprehensive set of performance measurements 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

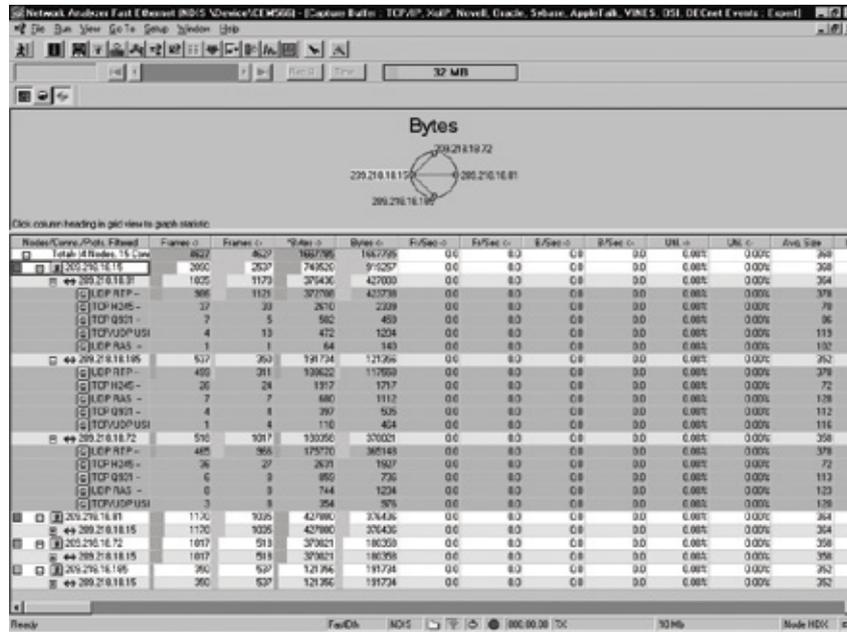


TNA CDR Screen

**Call Detail Records**

The VoIP Expert Commentators produce a record of all H.323 and SIP 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
- P and UDP addresses

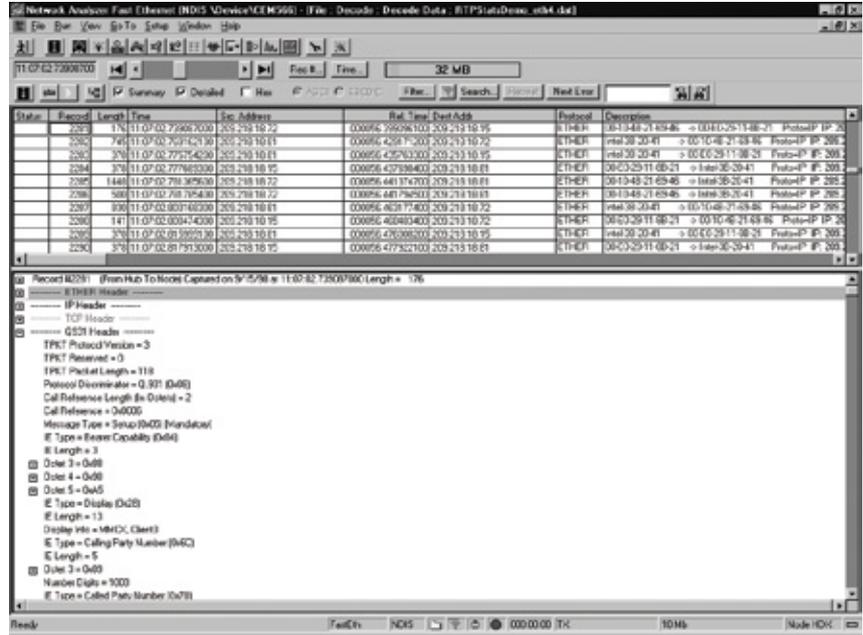


TNA connection stats screen

**Connection, Node and Protocols Statistics**

Counts for traffic levels for each Connection pair, (IP address pair) each node and 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, and can view protocol distribution for all addresses. 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. The user can also drill down to see the traffic levels for each application, by protocol, between the two workstations making up the connection.



RTP Decode Screen

**Protocol Decodes**

The JDSU Network Analyzer has the most extensive library available for VoIP protocol decodes. All VoIP standards including H.323, SIP, MGCP, SGCP, H.248/MEGACO, NCS, RTP, RTCP, SCTP, SIGTRAN, and more are decoded and presented in simple text summary, detail, and hexadecimal formats. Other data and fax protocols, including T.120 and T.38, are also supported.

**Automated Hierarchical Process to Troubleshooting**

So how do all of these features work together to provide simple, fast and powerful troubleshooting solutions?

The VoIP Expert Commentators detect and inform the user of errors and anomalies. Expert help text explains why the error may be a mission-critical problem. Should the user wish to view a detailed decode, one click will display the culprit packet or packet ranges. Similarly, the VoIP Expert Commentators may return an “Insufficient Bandwidth” warning. One 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 Telephony Network Analyzer 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 Telephony Network Analyzer 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 Telephony Network Analyzer'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.

### **The Telephony Network Analyzer Provides the Expertise**

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 Telephony Network Analyzer to ensure the rules of the H.323, MGCP and SIP standards are obeyed by their design. Compliance with the standard will assist interoperability when that product is connected to other vendor's products. 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 "noncompliant" 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 non-compliant behavior, but is not necessarily catastrophic. However, it is inefficient and may cause problems with other vendors' equipment.

The JDSU 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.

Call Control of voice or video conferencing over IP contains complexity at every stage. The VoIP Expert Commentators on the Telephony Network Analyzer offer 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.

### **Reporting**

Results from the Telephony Network Analyzer can be quickly and easily compiled into professional reports for management and customer presentation, or for performance trending analysis. The JDSU Report Center is a PC software application that provides instant visualization of QoS on VoIP networks with innovative graphs of network performance and service quality parameters. It correlates measurement results from the JDSU Voice Quality Tester (VQT), Telephony Network Analyzer, and Network Analyzer to give a simple view of network performance in terms of voice clarity, delay, echo, packet loss, jitter, line utilization, and other key network metrics.

### **Telephony Network Analyzer Key Features**

- Automatically detects VoIP calls, and performs real-time analysis on VoIP calls and endpoints
- Measures and analyzes RTP packet loss, jitter, and counts for each call and for each endpoint
- Predictive MOS (non-intrusive voice quality measurement)
- Works over any LAN/WAN technology up to Gigabit Ethernet and OC-12/STM-4, including MPLS and IPv6 networks
- RTCP and call statistics monitor, including packet loss, jitter, delay measured by the VoIP end points
- Real-time audio playout of VoIP packet streams, for G.711, G.729a/b, and G.723.1 codecs, with and without jitter buffering
- Simplified troubleshooting of call signaling and control through embedded expert analysis of SIP, MGCP and H.323 protocols.
- Call Detail Records for each call or part of call, with automatic references to signaling anomalies and specifications
- Provides results in graphical and numerical presentations
- Identifies RTP packets out of sequence
- Automatic drill-down to packet-by-packet analysis and RTP decodes
- Alarms on user-defined conditions
- Real-time decodes, filtering, and analysis on all VoIP protocols for LAN and WAN networks.
- Exports results in CSV format to spreadsheet

**Applications**

- Certify and qualify new VoIP deployments
- Troubleshoot VoIP service quality impairments
- Troubleshoot VoIP signaling and control
- Baseline network performance
- Monitor network performance and SLAs
- Optimize VoIP networks and services
- Optimize voice quality on the network via configurations of jitter buffers, codec and silence suppression selection, and network routing, QoS and queuing. Measure the results of each configuration with the Telephony Network Analyzer to determine the optimal balance of voice quality with network bandwidth utilization.

**Platforms**

- The JDSU Telephony Network Analyzer is a software solution available on all JDSU Network Analyzer platforms:
- J6800A Network Analyzer mainframe
  - J6801A Distributed Network Analyzer
  - J6802B Distributed Network Analyzer MX
  - J6805A Distributed Network Analyzer ME
  - J6840A Network Analyzer Software
  - J6839A Network Analyzer Software Professional Edition

**VoIP Protocol Analysis**

Available on Network Analyzer and JDSU Advisor standard packages:

H.225	RTCP
H.235	RTP/c-RTP
H.245	SAP
H.248 MEGACO	SDP
H.261	SGCP
H.263	SIGTRAN (IETF SS7/IP)
H.323	SIP
H.450.1	SIP-T
H.450.2	T.120
H.450.3	T.122
MGCP NCS (PacketCable)	T.123
SCTP	T.125
Q.931	T.38
RAS	

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**Notes**

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