# the MediaPro

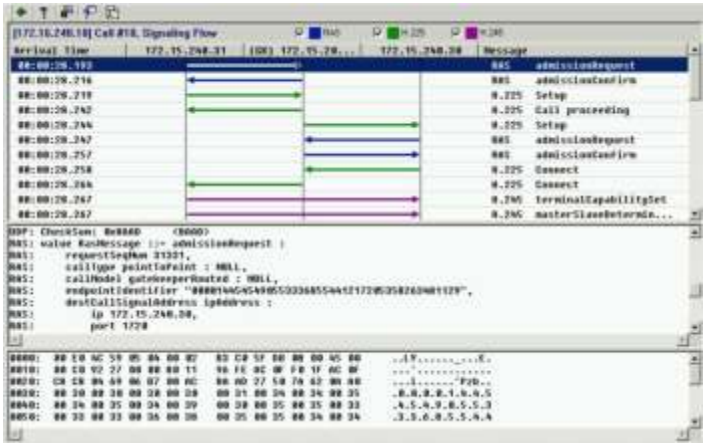
## Online, Session-oriented IP Multi-media Analyzer

The MediaPro™, a cutting-edge, high-performance, session-oriented IPMM (IP Multi-media) analyzer, provides accurate problem detection, fast troubleshooting and IP network performance analysis.

Intended for the QA Engineer testing high volume next generation architectures and solutions, or for the engineering team in charge of deployment and maintenance of the increasingly loaded IPMM live network, the MediaPro provides complete signaling and media analysis, including QoS, video and voice quality. It detects problems such as inefficient bandwidth utilization, inefficient packet loss recovery mechanisms, poor compression mechanisms, non-optimal jitter buffer length, quality degradation under stress, inadequate VAD (Voice Activity Detection) mechanisms and long signaling setup durations. In addition, full drill-down capabilities enable isolating every level of the signaling and media planes.

The MediaPro associates signaling and media per call-base, and provides a full view of the calls executed by VoIP/PSTN devices in the VoIP network. It is the ultimate solution for IPMM troubleshooting, debugging and regression tests.

The MediaPro provides high performance session analysis using dedicated hardware, based on the GEAR (GEneric AnalyzeR), RADCOM's processor chip.



*Call signaling flow*


## Highlights

- Allows real-time IPMM network analysis.
- Provides high performance capture capability.
- Uses dedicated hardware, based on RADCOM's GEAR chip.
- Connects to 10/100/1000 Mbps Ethernet and other networks.
- Associates signaling, voice and video streams per call base.
- Provides an embedded jitter buffer consultant.
- Evaluates non-intrusive voice quality monitoring based on the industry standard ITU-T G.107 E Model (MOS and R-Factor evaluation).
- Evaluates objective voice quality using PESQ (ITU-T P.862) and PAMS.
- Supports multi-IPMM signaling, including SIP, H.323, MGCP, Megaco, SCCP, NCS, TGCP, SIPT, C-SIP and more.
- Calculates enhanced jitter and inter-packet delay variation using an automatic expected packet length mechanism.
- Allows audio transparent playback taking into account jitter, packet loss, silence suppression and packet order.
- Allows video transparent playback taking into account jitter and packet loss.
- Provides statistic reports and graphs.
- Provides enhanced hardware filtering capability including:
  - Protocol filtering.
  - Source and destination IP addresses.
  - Calling party number, called party number.
  - Closed call filter of signaling failed calls.
- Provides automatic test capabilities through the MasterScript, a powerful scripting tool.
- Provides analysis of cRTP traffic.

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