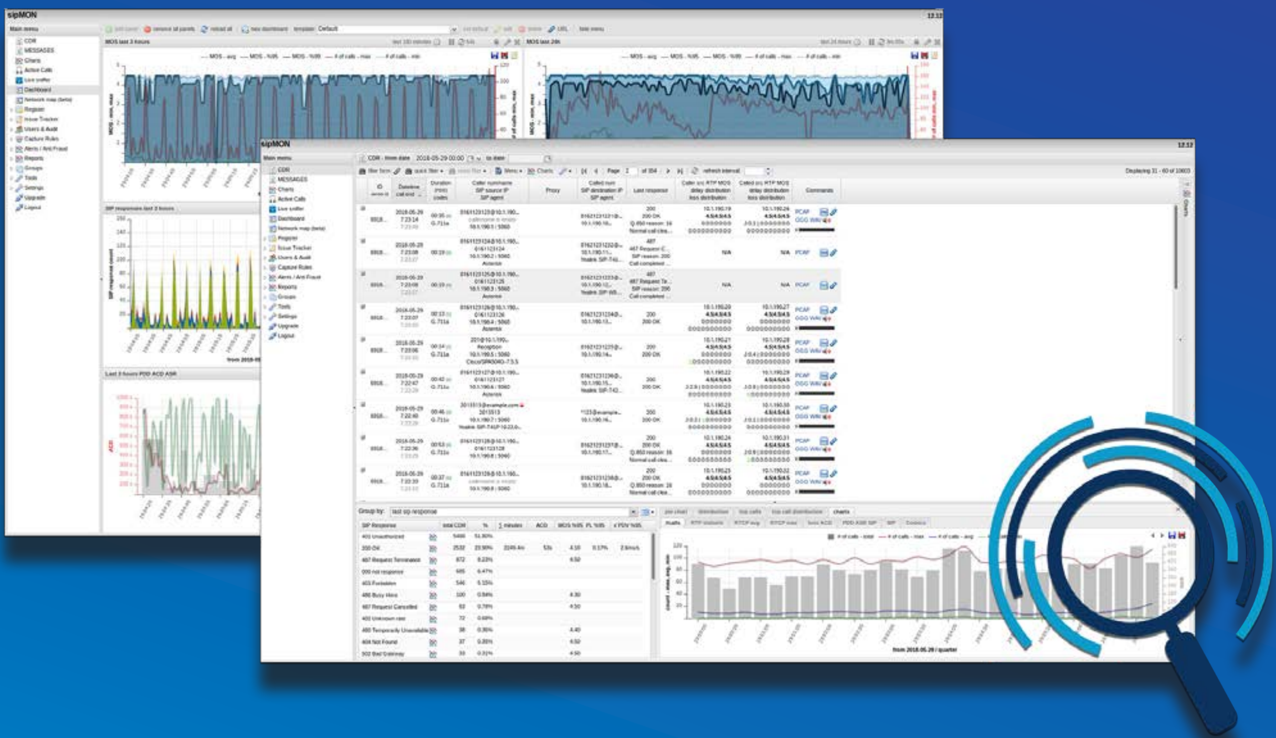
BICOM SYSTEMS
ADVANCED SIMPLICITY

SIP MONITORING

analyzing every packet of every call



sipMON

PRICELESS VISIBILITY OF CLIENTS` NETWORK

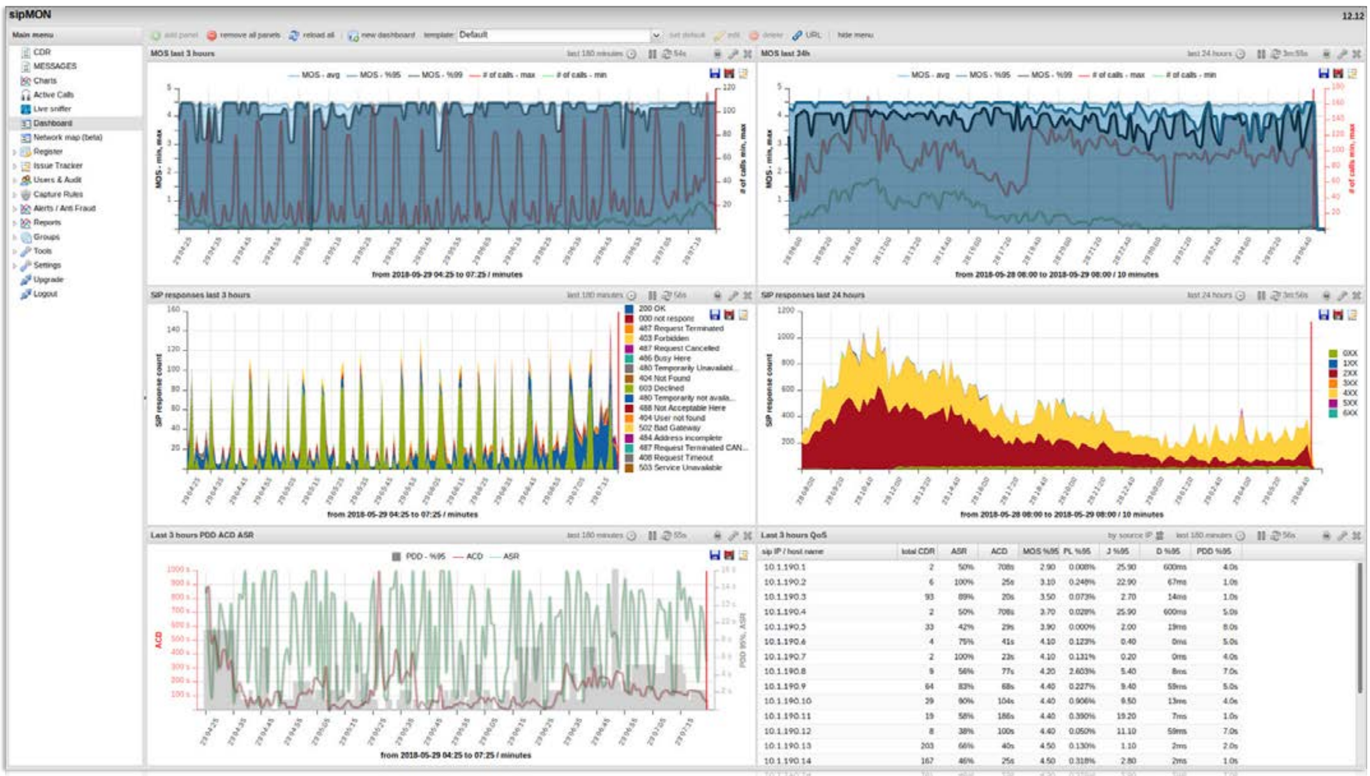
bicom SYSTEMS

sipMON

sipMON is a network packet sniffer for SIP and RTP VoIP protocol specifically designed to work with PBXware.

sipMON is able to handle thousands of simultaneous calls. It listens on a network interface and analyzes all SIP calls on defined SIP ports. RTP streams, which carry voice, are analyzed for packet loss and variation delay (jitter).

Each call is saved to the database supporting ODBC. SIP signalization and RTP packets are saved to an individual pcap file which can be opened with analyzers such as sipMON GUI.



KEY FEATURES

Jitter Monitoring

sipMON allows monitoring of relevant jitter data for all calls. It uses a jitter buffer simulator to keep both directions of calls synchronized.

Delay Monitoring

Show variable delays delimited by ‘:’. The first number is the number of delays between 50-70ms, the second is between 70-90, next is 90-120, 120-150, 150-200, 200-300, 300-more.

Packets Transfer Monitoring

Display lost packets distribution delimited by ‘:’. The first number counts the loss of one isolated packet. The second is two consecutive lost packets, next is 3, 4, 5, 6, 7, 8, 9 and 10-infinite lost packets.

Call Recording

sipMON automatically records all phone calls established over the users’ PBXware. sipMON can also decode speech and play it over the sipMON GUI or save it to a disk as WAV.

MOS Score

Mean Opinion Score. There are three MOS values: Fixed 50 – simulated jitter buffer for devices with almost no jitter buffer, Fixed 200 – simulated jitter buffer for devices with 200ms fixed jitter buffer, Adaptive 500 – Simulated jitter buffer for devices with the adaptive 500ms jitter buffer.

Data Transfer

Call data is automatically saved to a pcap file with either only SIP protocol or SIP/RTP/RTCP protocols. Files may be exported to a hard drive at any moment. Calls with all relevant statistics are saved to the sipMON database.

Live Calls

Real-time monitoring of ongoing phone calls. This feature requires the latest version of sipMON with an enabled TCP manager port.

RTP Monitoring

sipMON displays a diagram of RTP stream from all IP addresses, callers, and call receivers. RTP stream diagrams are separated for both sources.

HOSTED or ON-PREMISE

At Bicom Systems, we offer both hosted and on-premise solutions for Unified Communications. When you choose the hosted deployment option, you are relieved of the burden of purchasing and maintaining the hardware. Companies that opt for the on-premise solution have the complete control over all their servers and data.

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to find out more about our services

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