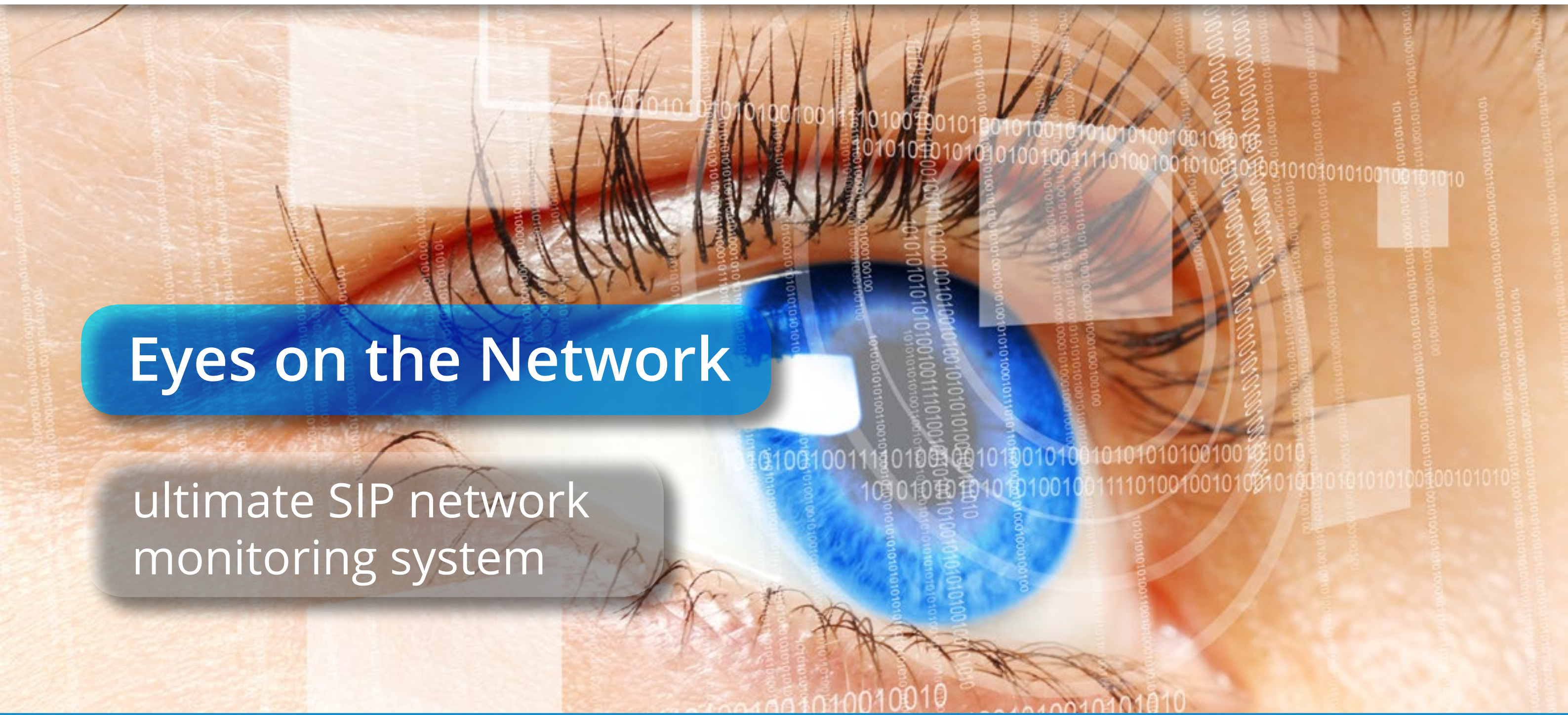
bicom SYSTEMS

sipMON

SIP Monitoring System



Benefits

• Rapid response
React to the network problems and solve them before your customers even realize there is something wrong.

• Better customer service
Provide stable quality service to your customers by preventing service failures and system downtime.

• Prevent financial loss
System downtime and low quality of service tends to greatly reduce ISPs income and customer base.

• 24/7 monitoring
SIP network will never be unattended and exposed to risks. The world does not sleep - sipMON doesn't either.

• Save time
Your support tehnicians and network administrators can focus on solving rather than locating the issue.

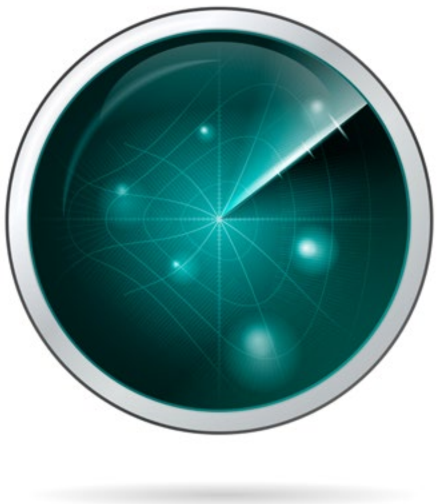
• Increase revenue
Quality service is highly appreciated and presents a best form of advert able to generate a great revenue.

General info

sipMON is the essential tool for any serious ITSP. It is a network packet sniffer for SIP and RTP VoIP protocol developed to work with our PBXware.

sipMON was designed to analyze quality of VoIP calls based on network parameters like Jitter, Delay variation, Packet loss according to ITU-T G.107 E model, Predicts quality of calls on MOS scale, etc.

Enables providers to monitor their network 24/7 and respond to any eventual issue in shortest amount of time, preventing escalation.



Key features

- Live SIP Packet Sniffer
- Selectable Sniffing Sensors
- SIP Device Register Status
- Real Time Fraud Alert
- Jitter Monitoring
- Delay Monitoring
- RTP Monitoring
- Mean Opinion Score
- Comprehensive CDRs
- Call Recording
- Online Call Listening
- Save RTCP to PCAP file

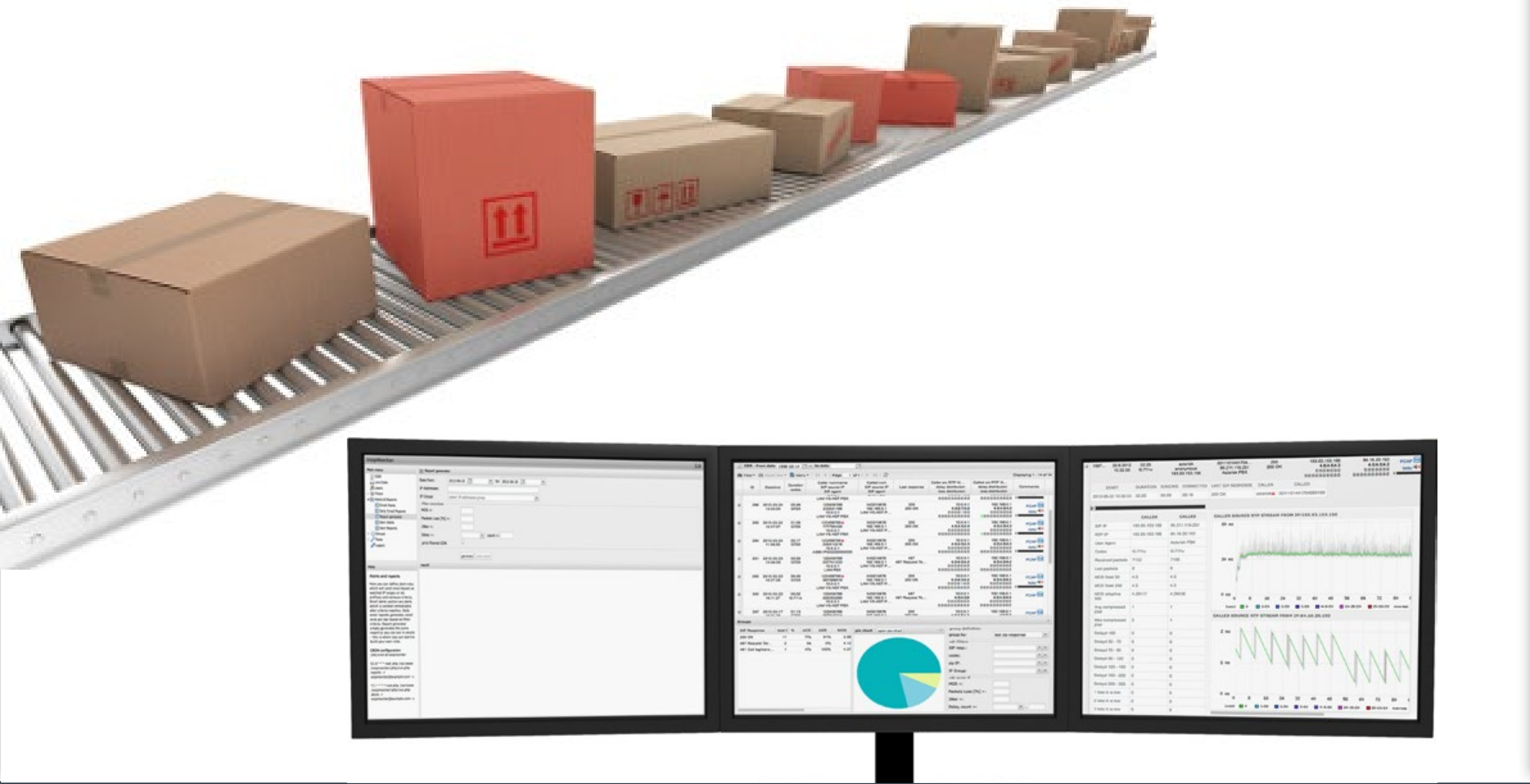
How it works

sipMON is one of the best SIP monitoring systems available, designed to handle thousands of simultaneous calls. It uses advanced detection systems to listen 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 log is saved to supporting database.

SIP signalization and RTP packets can be saved to an individual pcap file which can be opened with other analyzers like wireshark and is also used by sipMON GUI.

Resouce required: Dedicated Host Server



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Jitter Monitoring

sipMON allows monitoring of relevant jitter data for all calls. It uses jitterbuffer 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

Show lost packets distribution delimited by ':'. The first number counts 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.

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.

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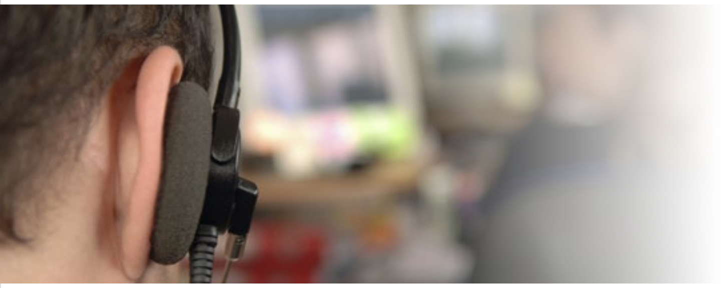
MOS Score

Mean Opinion Score. There are three MOS values: Fixed 50 | Fixed 200 | Adaptive 500.

- Fixed 50: Simulated jitterbuffer for devices with almost no jitterbuffer (max 50ms).
- Fixed 200: Simulated jitterbuffer for devices with 200ms fixed jitterbuffer.
- Adaptive 500: Simulated jitterbuffer for devices with adaptive 500ms jitterbuffer.



LIVE Calls Monitoring

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

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 the disk as WAV.

Data Transfer

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

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Comprehensive CDRs

CDR (Call Detail Record) contains call data and network statistics for every single call that users made.

CDR provides the following data organized into columns:

- ID: Unique autoincrement identification of calls. It is created on SQL INSERT.
- Datetime: Start of a call.
- Duration: Total length of a call from start to end.
- Codec: Audio codec used in a call.
- Caller num/name: Caller number and name from SIP header.
- SIP agent: Agent string from SIP header.
- Last response: Last SIP response, number and full text description.
- Caller/Called src RTP: Source IP address of incoming RTP packets from caller or receiver.
- MOS: Mean Opinion Score.
- Delay distribution: Show variable delays.
- Loss distribution: Show loss packets distribution.
- Commands

<table>
<tr><th>Last response</th><th>Caller src RTP M...
delay distribution
loss distribution</th><th>Called src RTP M...
delay distribution
loss distribution</th><th>Commands</th></tr>
<tr><td>200
200 OK</td><td>193.93.153.105
4.5|4.5|4.3
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>62.239.233.238
4.5|4.5|4.5
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>PCAP
WAV</td></tr>
<tr><td>200
200 OK</td><td>62.239.233.238
4.5|4.5|4.3
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>193.93.153.103
4.5|4.5|4.3
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>PCAP
WAV</td></tr>
<tr><td>484
484 Address inc...</td><td>N/A</td><td>N/A</td><td>PCAP</td></tr>
<tr><td>484
484 Address In...</td><td>N/A</td><td>N/A</td><td>PCAP</td></tr>
<tr><td>484
484 Address In...</td><td>N/A</td><td>62.239.233.238
4.5|4.5|4.5
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>PCAP</td></tr>
<tr><td>200
200 OK</td><td>65.254.43.163
4.5|4.5|4.3
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>193.93.153.103
4.5|4.5|4.5
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>PCAP
WAV</td></tr>
<tr><td>200
200 OK</td><td>193.93.153.103
4.5|4.5|4.3
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>62.239.233.238
4.5|4.5|4.5
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>PCAP
WAV</td></tr>
<tr><td>404
404 Not Found</td><td>N/A</td><td>N/A</td><td>PCAP</td></tr>
<tr><td>404
404 Not Found</td><td>N/A</td><td>N/A</td><td>PCAP</td></tr>
<tr><td>484
484 Address In...</td><td>N/A</td><td>62.239.233.238
4.5|4.5|4.5
0:0:0:0:0:0
0:0:0:0:0:0:0:0:0:0</td><td>PCAP</td></tr>
<tr><td>484
484 Address inc...</td><td>N/A</td><td>N/A</td><td>PCAP</td></tr>
</table>

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Fraud Alerts

sipMON monitors all phone calls picking up any fraudulent activities from sniffer and notifying administrator in real-time, even during the call.

LIVE Sniffer

Sniffing directly on PBX / Soft switch Linux or on dedicated Linux server mirroring traffic from switch.

Widget based Dashboard

Dashboard has a widget style design allowing easy customization of interface according to the user preferences. The charts are also configurable with implemented templates.

SIP Device Registration

sipMON monitors each device registration process and notifies admin on each of the devices status. Displays all SIP devices that are registering, failed to register or state of the registered device.

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Vision Statement

We Unify Communications!

Mission Statement

We provide the Communication World with the most Complete Turnkey Communication Systems available by Creating, Unifying and Supporting the Most Advanced of Current Technologies.

Overview

Bicom Systems was the first company to deliver Open Source Communications Software as Professional Turnkey Solutions.

By combining the best of open source telephony and its own proprietary software, Bicom Systems can provide enterprises with turnkey solutions that take account of the clients' exact needs within a very cost-effective framework - giving CIOs the safest choice. This mix includes royalty-free software, vibrant open source communities, available custom development backed up by accountable, professional support services.

The company finds innovative open source communication projects and professionalizes the project by creating, unifying and supporting turnkey systems with its proprietary in-house software. Bicom Systems provides the resources, core development and support services to enable popular open source projects to scale into enterprise-class communications software.

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