VOIP MONITORING – FINDINGS

KEY FINDINGS FOR VOIP MONITORING

2.88 Mean opinion score (MOS)	• Minimum MOS score observed for RTP provides insight into service level violations. MOS ranks from 1 to 5 with 1 being the worst.
9 milliseconds	• RTP jitter is acceptable, with the maximum jitter reaching only 9ms. Excessive jitter makes calls unintelligible.
2,800 SIP 401 status codes	• Responses with the 401 status code indicate unauthorized activity and should be investigated.
402 SIP “bad event from client” errors	• Call initiations that failed due to “bad event from client” errors. Users could not make calls.

INDUSTRY FACTS

- **Packet capture** is the most relied upon troubleshooting method for VoIP issues
  – Cisco support forum, 2014
- Voice was ranked as the **second-most used communication method** (86%, behind email at 93%) for employees
  – InformationWeek Reports
- **68% of consumers would hang up** as a result of poor call quality and call a competitor instead
  – Customer Experience Foundation

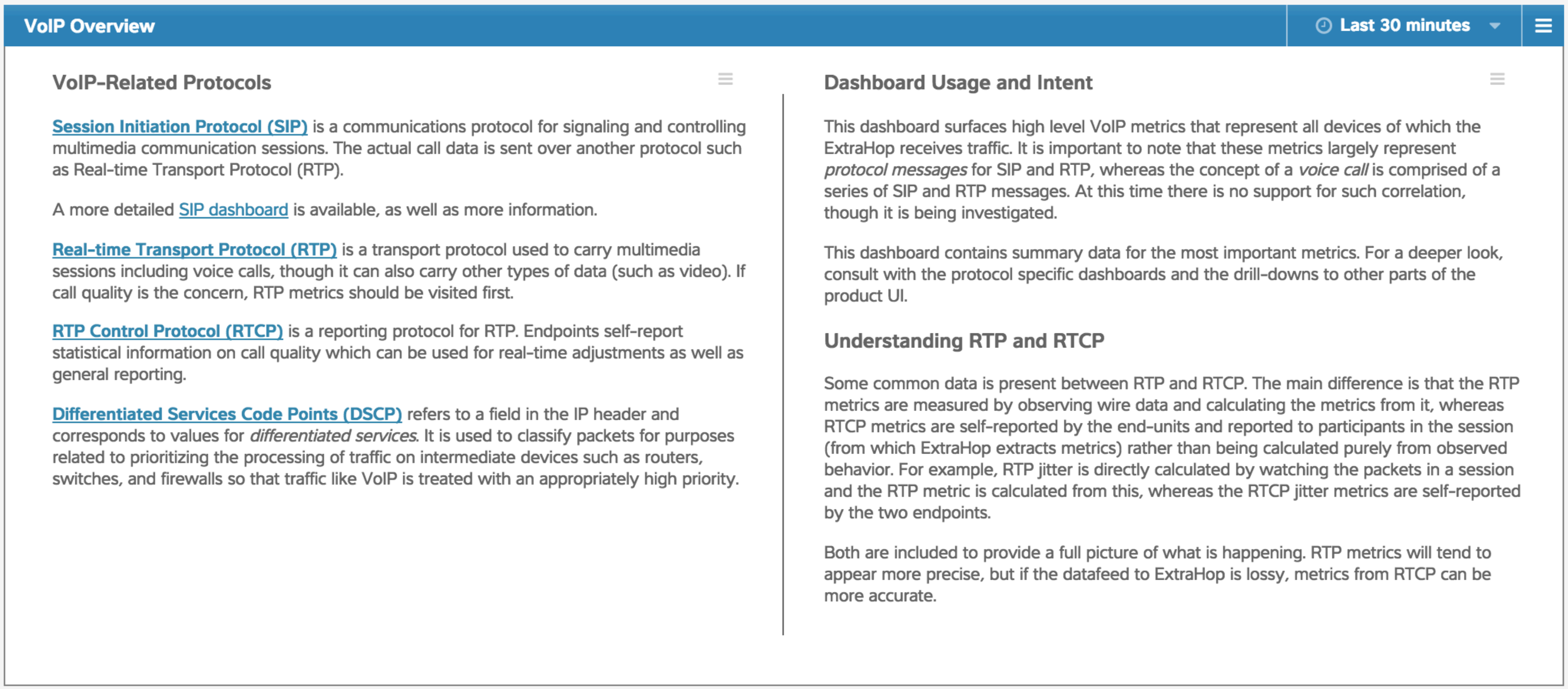
VOIP MONITORING DASHBOARD

The VoIP Monitoring dashboard surfaces high level VoIP metrics that represent all devices for which the ExtraHop receives traffic. The ExtraHop platform performs real-time analysis for call setup and control, and call quantity and quality for all VOIP sessions. In addition to an overview dashboard for all VoIP protocols, the project also includes a dashboard for observed detailed metrics for the Session Initiation Protocol (SIP) as well as for combined detailed metrics for Real-time Transport Protocol (RTP), RTP Control Protocol (RTCP), and Differentiated Services Code Points (DSCP).

Here you see the worst MOS scores and jitter from the observed period. MOS and jitter are the metrics used to determine call quality, so you should use this view to track VoIP service levels.

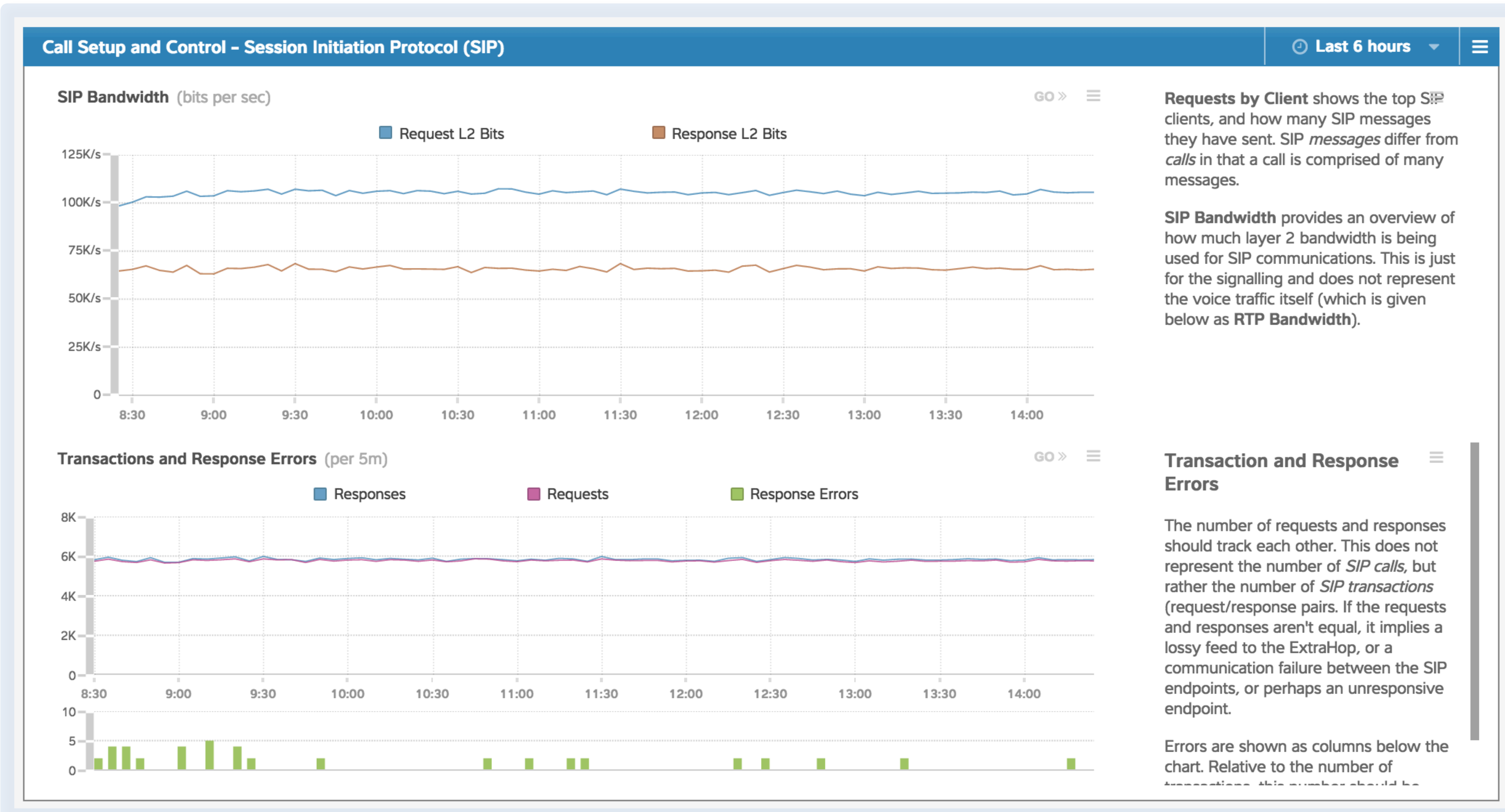
VOIP MONITORING DASHBOARD

The VoIP Monitoring overview dashboard also includes explanatory text concerning the purpose of the dashboard and the protocols that it displays metrics for. This section also contains explanations for and links to four sub-dashboards that drill down into each protocol activity.



VOIP MONITORING DASHBOARD

This section of the VoIP Monitoring dashboard tracks critical metrics for the Session Initiation Protocol (SIP), which controls call set up. These metrics are directly related to end users' ability to initiate VoIP phone calls and errors should be monitored closely.



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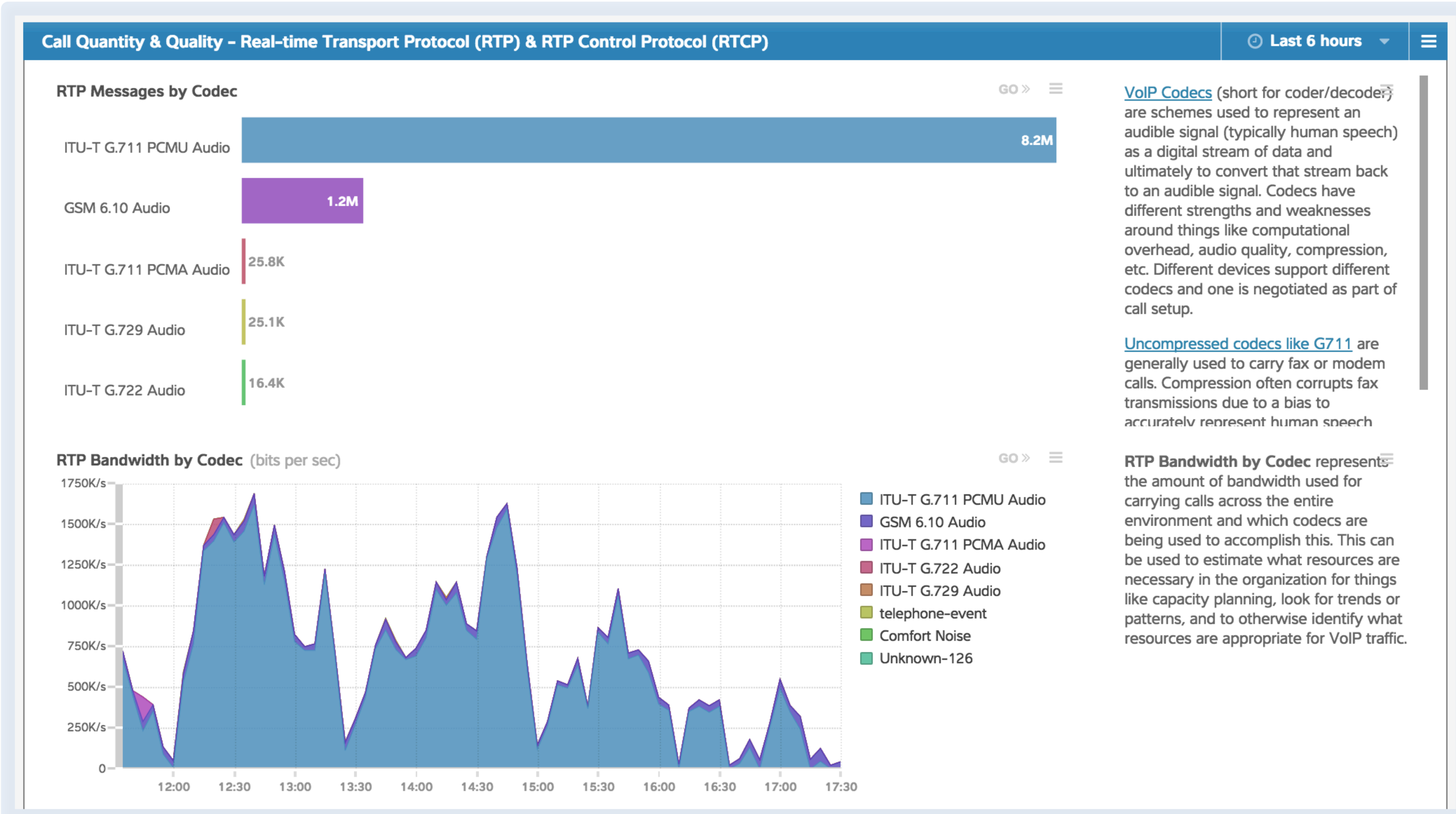
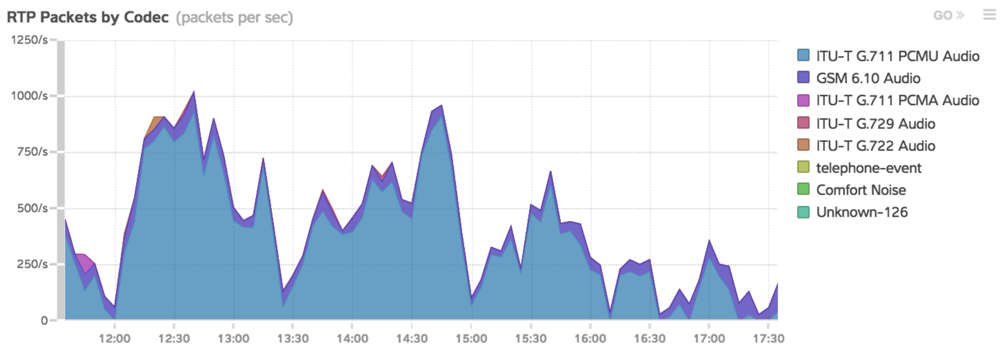
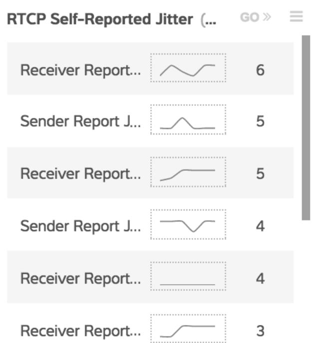
The section of the VoIP Monitoring dashboard shown below displays real-time information regarding Real-Time Transport (RTP) message codecs, which helps network teams understand the types of VoIP traffic that is passing over the network. This operational intelligence is crucial when monitoring the impact of network bandwidth and throughput on VoIP services and also for capacity planning. .

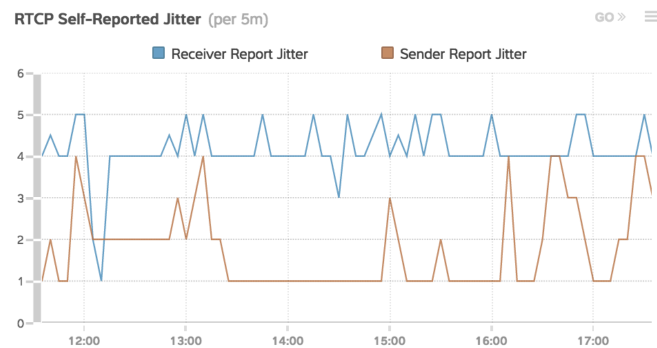


VOIP MONITORING DASHBOARD

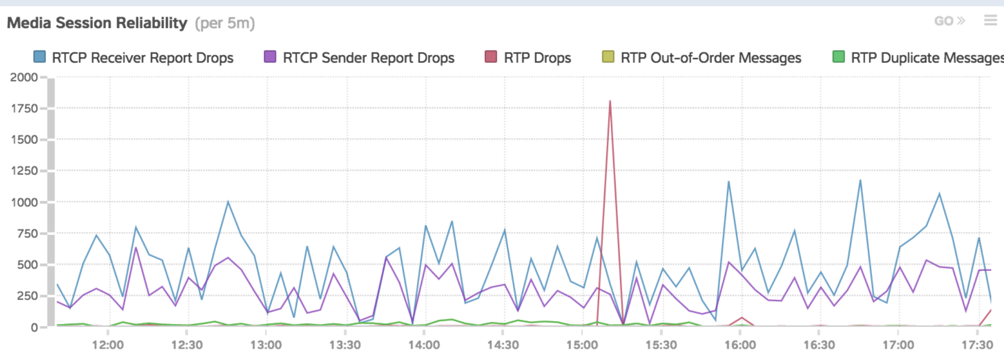


RTP Packets by Codec breaks down which codecs are in use the most by the number of packets used for their media streams. If this chart doesn't roughly track the same shape of the **RTP Bandwidth by Protocol** chart, it can be indicative of a particularly good or bad codec in use relative to the others (depending upon whether the number of packets per second is higher or lower than expected).



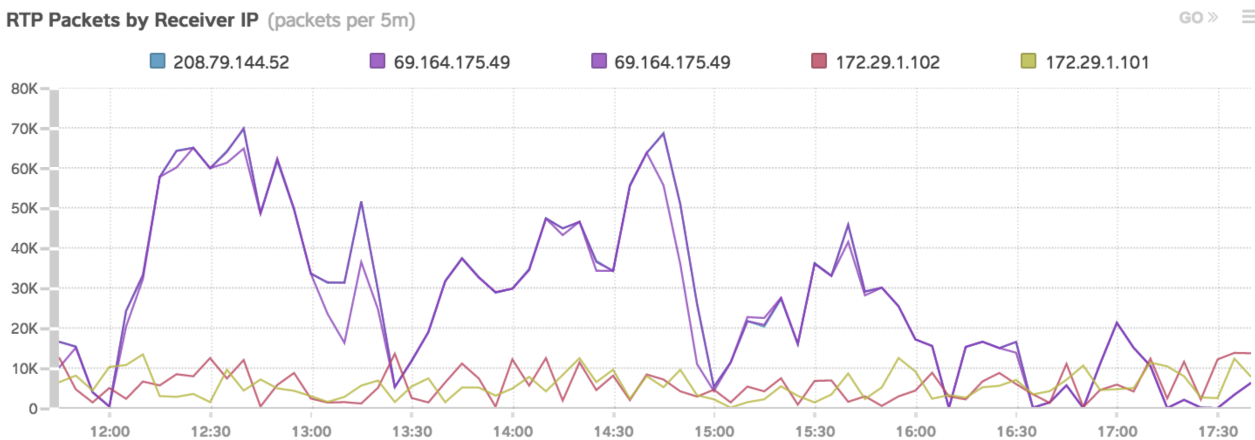


RTP Jitter (provided under *Critical Metrics*) represents observed behavior, whereas the **RTCP Sender and Receiver Report Jitter** metrics are self-reported by endpoints. Low values are better, and values over 100ms begin to impact call quality.



RTP can handle some drops and out-of-order messages without impacting call quality depending upon the codec, device buffering and the amount of such incidents, but drops and out-of-order messages are reason for concern. If you notice spikes in these metrics, or if users complain of call quality issues that correspond to elevated numbers in this chart, taking action to prevent these conditions is advised. Likewise, duplicate messages, while not fatal, can be indicative of a problem.

VOIP MONITORING DASHBOARD



RTP Packets by Receiver IP will show which end units are receiving the most traffic. This will often show similar results as **RTP Packets by Sender IP** but does not have to, and in such a situation can imply things like unidirectional conversations (simplex communications).

VoIP Codecs (short for coder/decoder) are schemes used to represent an audible signal (typically human speech) as a digital stream of data and ultimately to convert that stream back to an audible signal. Codecs have different strengths and weaknesses around things like computational overhead, audio quality, compression, etc. Different devices support different codecs and one is negotiated as part of call setup.

Uncompressed codecs like G711 are generally used to carry fax or modem calls. Compression often corrupts fax transmissions due to a bias to accurately represent human speech

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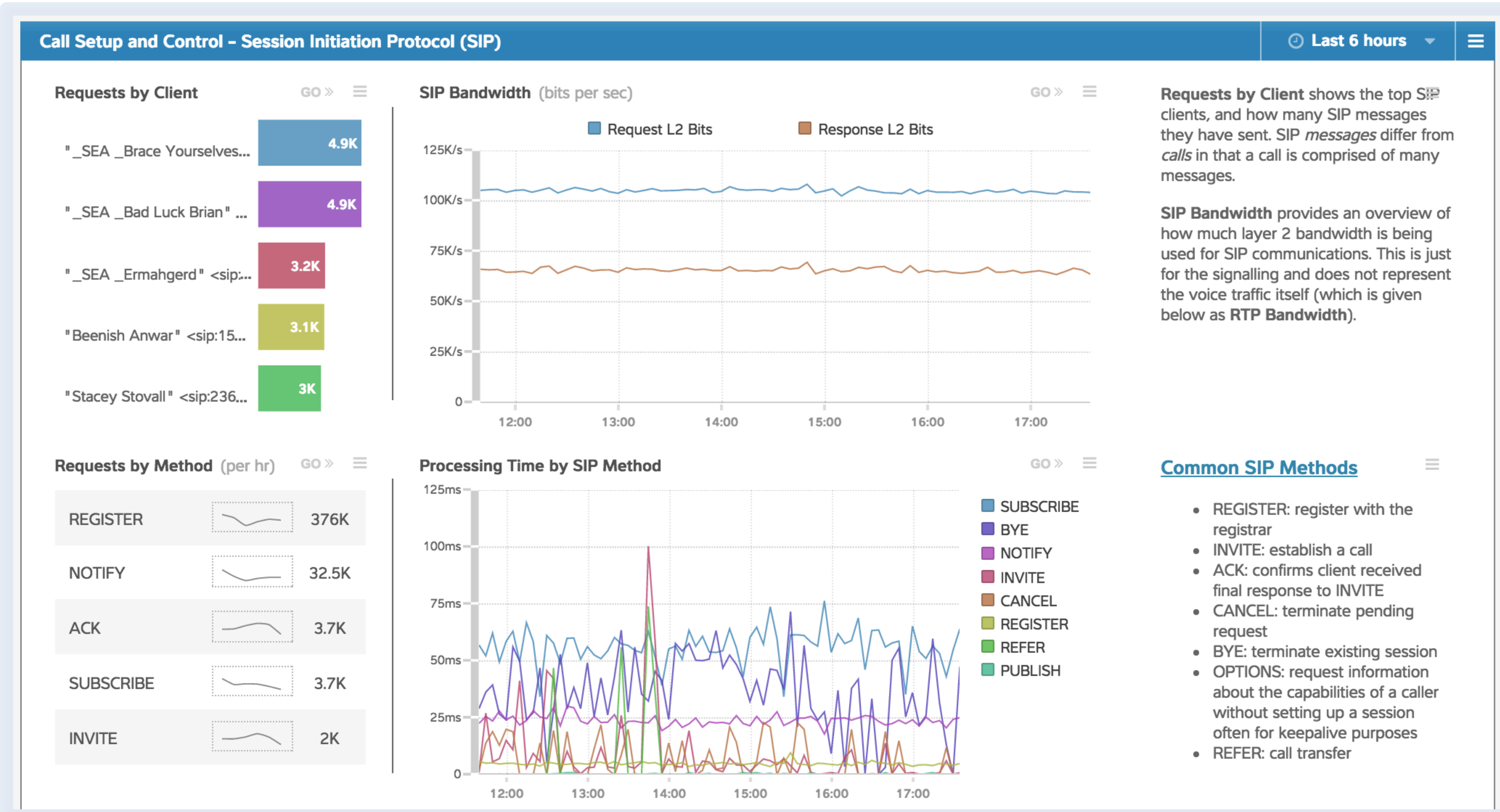
The SIP-specific dashboard focuses on call setup metrics. The Requests by Client shows which phones are making the most VoIP calls. The chart showing the processing time for each SIP method is helpful for network teams trying to quickly determine the cause of reported issues.



VOIP MONITORING DASHBOARD

The SIP-specific dashboard continues with counts of each code along with the processing time.