VoIP Analysis

T-BERD®/MTS-6000A & 8000 MSAM Ordering Information:	CTVOIP(-U1) – Includes SIP & H.323 CTSCCP(-U1) – Adds SCCP to above CUSB-HEADSET – Headset for audio
Use Case:	Verifying VoIP Services in Ethernet/IP networks

Work Groups and Applications

Central Office, Metro, and Government technicians responsible for VoIP service installation, services turn-up, and troubleshooting.

Engineers responsible for the maintenance, troubleshooting and evolution of Ethernet/IP networks carrying VoIP traffic.

Applications

Installing, verifying, and troubleshooting VoIP services supporting SIP, Cisco SCCP, and H.323 Fast Connect signaling protocols.

Verifying the suitability of Ethernet/IP packet networks to reliably transport VoIP traffic with audio/voice analysis and simultaneous background traffic.

Solution Description

The VoIP test options for the T-BERD/MTS-6000A & 8000 MSAM ensure the successful installation of VoIP services by emulating an IP phone and placing and receiving a VoIP call at all Ethernet rates up to 10G. By supporting SIP, Cisco SCCP, and H.323 protocols, connectivity to the signaling gateway can be verified. Technicians can quickly and objectively prove acceptable call quality with real-time analysis of audio delay, jitter, packet loss, MOS and R-Factor; all of which can be given a good, fair or poor quality rating with configurable QoS thresholds. Extensive troubleshooting capabilities include visibility into the entire call set-up signaling process and the support of line rate capture and decode of both audio and signaling packets.

Users can also verify the ability of a packet network to reliably transport voice traffic by utilizing the Triple Play Application to generate and analyze the QoS of an audio/voice stream with simultaneous background traffic. This will prove acceptable voice/audio quality under load conditions and expose potential high bandwidth utilization problems such as mis-configured prioritizations for voice traffic.

Value Proposition

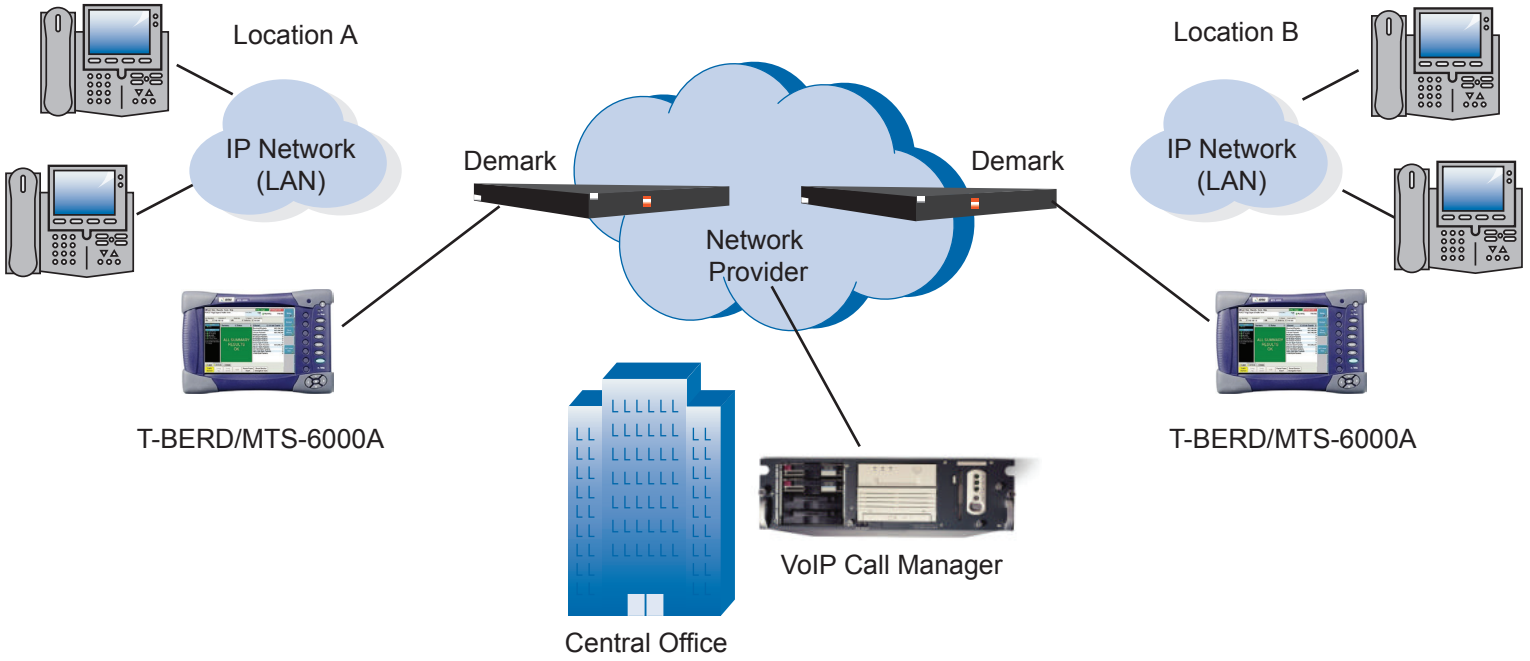
As more Time Division Multiplexing (TDM) circuits are converted to VoIP trunks, metro technicians require a tool to objectively verify VoIP call quality and to troubleshoot those services more quickly. By looking beyond the transport layer and emulating the true end-user VoIP service or experience, this software test option allows technicians to ensure end user satisfaction in a cost effective way. With easy to decipher pass(green)/fail(red) objective QoS analysis organized by layer of the network, technicians can quickly identify and sectionalize problems, saving crucial turn-up and troubleshooting time. Integrated troubleshooting tools such as capture and decode as well as a real-time signaling log eliminate the need for highly experienced technicians or separate VoIP analyzer equipment. The unique ability to analyze voice/audio QoS with background traffic in the Triple Play Application, allows users in one simple step to stress the circuit and confidently prove that Class of Service prioritizations are functioning properly.

Benefit/Feature Summary

Benefit	Feature	Description	Advantage
Verifies true end user VoIP call experience with real voice or pre-loaded audio clips and ensures satisfaction before enabling VoIP services	IP Phone Emulation with SIP, SCCP, H.323	Use the test set to place and receive a VoIP call with a choice of signaling protocols at all line rates from 10M to 10G	Fast and simple way to verify connectivity to a signaling gateway or proxy server
Guarantees that all configurations are possible in one tool with flexible analysis and ubiquitous usage	Wide Range of CODEC Support	Complete set of standard ITU-T CODECs including G.711 μ-law/A-law, G.723.1, G.726, G.729a, G.729ab, and G.722	Multiple choices are available, encompassing all network and equipment possibilities
Allows for fast identification and sectionalization of problems during service turn-up and troubleshooting	Layered Summary Results	QoS measurements are presented as pass/fail (green/red) organized by network layer (Physical, Ethernet/IP, RTP)	Provides repeatable and simple to understand results as well as detailed statistics
Ensures the service meets true customer QoS in a repeatable and consistent way	MOS and R-Factor Scoring	Patented Telchemy real-time assessment of voice quality	Using a standardized method of measuring call quality eliminates listener subjectivity
Integrated troubleshooting analysis in one easy to use tool eliminates the need for experienced technicians and additional analyzer equipment	Troubleshooting with Capture/Decode & Signaling Log	Line rate capture and decode of audio and signaling traffic as well as a real-time signaling log for call set-up	Quickly and efficiently troubleshoot call set-up problems with clear identification of errors
Ensures true quality of VoIP services and confirms proper prioritization of voice traffic within the transport network	Audio/Voice Analysis with Background Traffic	Simultaneous line rate generation and analysis of an audio/voice stream with background traffic	Exposes high bandwidth network utilization problems which can negatively impact voice quality

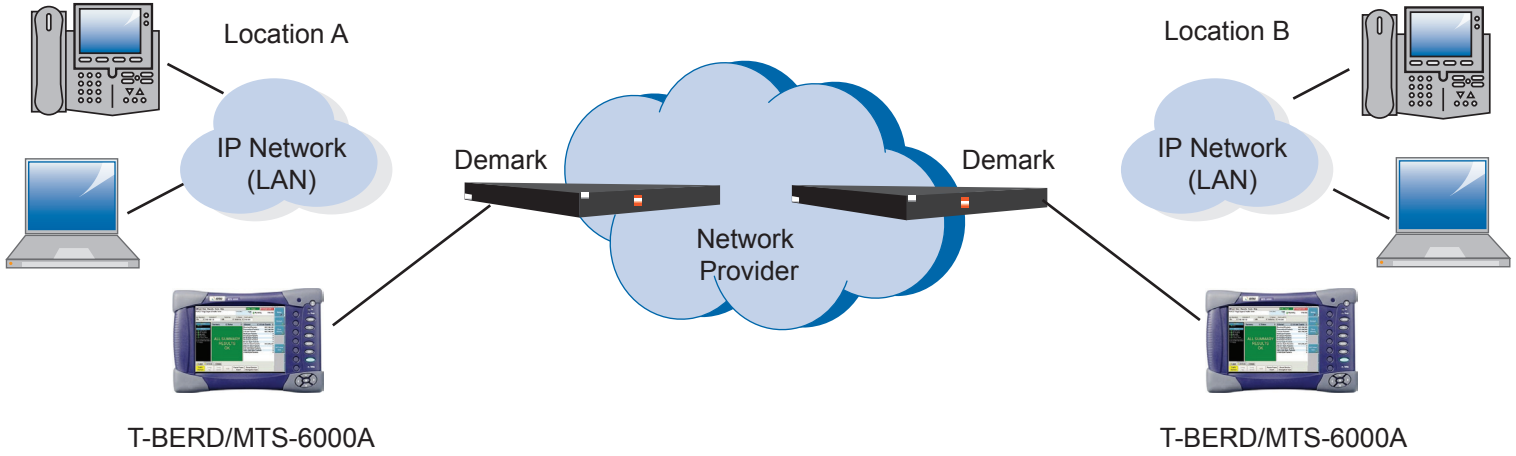
Use Case: VoIP Call Placement

The T-BERD/MTS-6000A & 8000 MSAM can be used to emulate an IP phone to place and receive a VoIP call through the network by calling another test set at the far end or any phone. In this scenario the test set is configured in terminate mode and is used to place a VoIP call to a remote IP phone by first connecting or registering with the VoIP call manager or proxy server. Real voice or a pre-loaded audio clip can be transmitted through the network and listened to by connecting a JDSU provided USB headset to the test set. Upon call set up, the test set analyzes audio/voice and transport QoS statistics in real time such as delay, jitter, packet loss, and MOS scoring, giving pass/fail indications.



Use Case: Triple Play Voice Analysis with Background Traffic

The T-BERD/MTS-6000A & 8000 MSAM can be used to generate and analyze voice traffic with simultaneous background traffic at line rate. This allows for the verification of a packet Ethernet/IP network to reliably transport voice along with data and does not require a VoIP call manager or signaling. In this scenario the test set is configured in terminate mode using the Layer 3 or Layer 4 Triple Play Application, which allows for the generation of multiple traffic flows including Real Voice, Simulated Video, and Data. Different priorities and settings can be configured per traffic stream and key SLA/KPI parameters such as delay, jitter, and loss are also measured per stream. The pass/fail QoS measurements for voice/audio are analyzed while the transport pipe is filled with background traffic, exposing any negative impacts that high bandwidth utilization will have on voice quality.



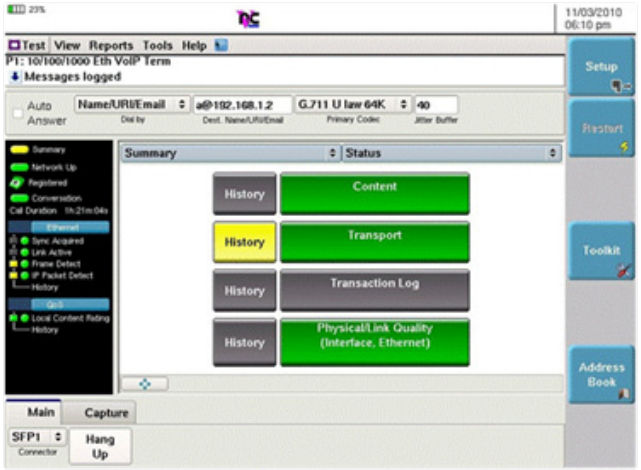


Figure 1: Layered Summary Results

Simplified VoIP Results

Call summary results are presented in a simple easy-to-read manner by network layer allowing quick identification of where in the network the problem is occurring. (Figure 1)

All transport call QoS measurements are viewed in a single table. Pass/fail indications are instantly revealed and can be saved into a report for proof of verification. (Figure 2)

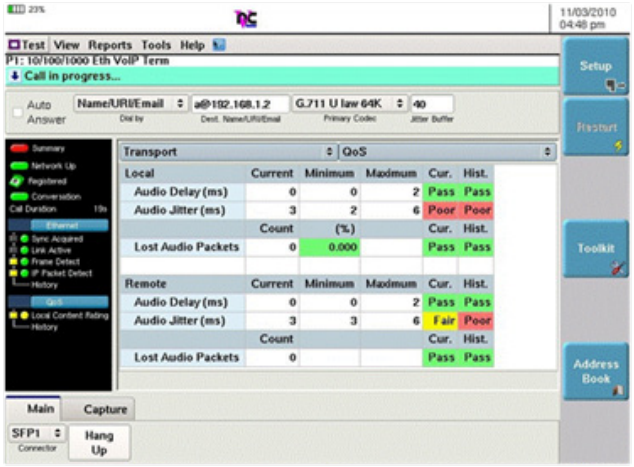


Figure 2: QoS Stats with Pass/Fail

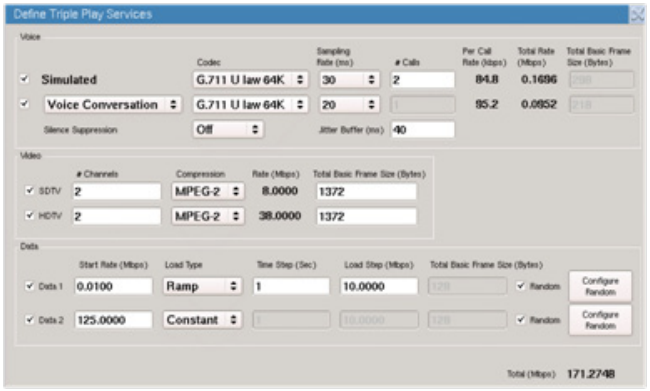


Figure 3: Triple Play Services Configuration

Simplified Triple Play Set Up and Results

Triple Play traffic streams configuration is optimized to remove the complexity of emulating multiple services and is flexible in allowing different set-ups per stream. (Figure 3)

QoS results and errors per stream can be observed in a transport pipe view indicating simple pass/fail, or in a graphical format over time. (Figure 4)

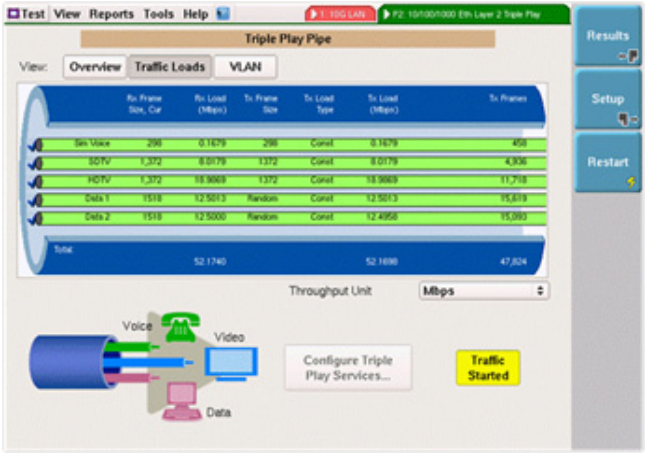


Figure 4: Triple Play Results Pipe

FAQ

Q: Which signaling protocols and CODECs does this option support and on which interfaces?

A: The T-BERD/MTS-6000A & 8000 MSAM VoIP options support SIP, Cisco SCCP, and H.323 Fast Connect signaling protocols on all Ethernet interfaces from 10M up to 10GE. Supported CODECs include G.711 µ-law/A-law, G.723.1, G.726, G.729a, G.729ab, and G.722.

Q: What is the advantage of using a tester for VoIP instead of just listening to a call?

A: The test set has in depth statistics that can be used to troubleshoot registration issues with SIP, SCCP, or H.323 fast connect protocols that cannot be accomplished with a phone. It also provides a qualitative MOS score that removes listener subjectivity and impartially determines whether the call quality is truly good or bad.

Q: Can I capture the voice call and playback later?

A: The test set supports capture and decode of both the voice call and the signaling packets. Filters can be used to capture a specific call or to capture just voice or signaling packets. The test set does not support playback at this time.

Q: Is this a software or hardware upgrade to existing units in the field?

A: The VoIP option is a software upgrade that applies to all MSAM hardware configurations.

Q: Are there pre-requisites for this feature?

A: No, there are no pre-requisites. A JDSU supplied USB headset is required to generate and hear voice/audio. Part number CUSB-HEADSET.