

DSAM VoIP Offerings App Note

DSAM VoIP Offerings

DSAM VoIP



JDSU

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1 Acronym List

- DSAM – JDSU’s Digital Service Activation Meter
- CMTS – Cable Modem Termination System
- CM – Cable Modem
- DOCSIS – Data-Over-Cable Services Interface Specification
- MER – Modulation Error Rate
- MTA – Multimedia Terminal Adapter
- CPE – Customer Premise Equipment
- HSD – High Speed Data
- VoIP – Voice over Internet Protocol
- HFC – Hybrid Fiber Coax
- MG – Media Gateway
- QoS – Quality of Service
- MOS – Mean Opinion Score
- CODEC – Compression-Decompression
- eMTA – Embedded MTA
- CMS – Call Management System
- NCS – Network-based Call Signaling
- DQoS – Dynamic QoS
- UGS – Unsolicited Grant Service

2 Introduction

2.1 Background

Voice over IP has long been touted as the “next big thing” in the future of the CATV industry. Ladies and gentlemen, the future is here!! Cable operators are drawn to voice services for their revenue potential, both in additional services, and in reduction of customer churn. Customers are drawn to it from a price and convenience perspective. Cable operators that ignore the industry shift into the voice service era will likely miss out on tremendous revenue opportunities.

The cable operators have offered High Speed Data (HSD) services for several years. They have become accustomed to delivery requirements of this service, and have successfully acquired and retained HSD customers. Voice is a different ballgame, with a whole new set of expectations and requirements. The major competition for the HSD system has typically been the dial-up system. Basically, there is no comparison between HSD and Dial-up. There is no question that HSD is a technically better solution, defeating dial-up in both performance and reliability. Customers compare their HSD connection to their old dial-up and are extremely satisfied with the HSD service. Voice, on the other hand, faces more stiff competition. The telephone systems that are in place today are unbelievably reliable. The customer likely will not even remember the last time that the telephone service was interrupted at their home. Even when the electricity was out, the customer was able to place a telephone call to report the outage. Cable operators choosing to deploy voice services do not have the analog telephone lines that traditional telephone companies have had running for decades. Instead cable operators have chosen to go with an all digital service in the form of VoIP for all their voice communication services. This different technology operates differently from traditional telephone service, but must perform as well, if not better, in order to maintain customer satisfaction. For these reasons, VoIP delivery will require a new level of testing and monitoring initiated by the cable operator.

To effectively deliver VoIP services, it is imperative that the proper test equipment be employed to test the cable network. The JDSU DSAM family of test instruments is designed to test cable networks. Test capabilities range from physical layer testing all the way to service layer testing, including voice testing.

The DSAM test instrument platform offers two tiers of VoIP test capabilities: VoIPCheck and PacketCable VoIP. Both of these tests have specific purposes, and targeted tests. There are advantages that each test has over the other. The following sections will explore these advantages.

3 DSAM VoIP Offerings:

3.1 VoIPCheck:

3.1.1 Overview:

The VoIPCheck measurement is intended to use the DOCSIS channel to assess the capability of the current test point to carry voice traffic. The intent is to perform a VoIP measurement without requiring the user to go through the extensive process of MTA provisioning. This test is ideal for customers that either do not use PacketCable™ to deliver VoIP (i.e. SIP or Vonage), or customers that have not yet partnered with JDSU to validate the operability of the DSAM Embedded Multimedia Terminal Adapter (eMTA) within their VoIP architecture.

3.1.2 Test Description:

The VoIPCheck test is broken into two parts. First, the DSAM can make the VoIP measurement to the first router in the pipe (Typically the CMTS). This will test the HFC portion of the network from the CM to the CMTS and back. This test is referred to within the DSAM as the CMTS Loop test. This test is practical if the goal is to determine whether the HFC drop can support a VoIP install. The packets that are exchanged are actual VoIP packets. This is a very important distinction, as the CMTS can de-prioritize packets that are deemed unimportant. If, for instance, 'Ping' packets were used in the same manner the resultant test may not be representative of the actual performance of a VoIP install.

Additionally, the customer can choose to install the JDSU Advanced IP Server near the MG (Media Gateway) in the system. In unison with the above CMTS Loop test, the DSAM can perform similar testing to this server. This view is known as the Segmentation test. The result of this test determines the quality of the entire path to the gateway. Using some intelligent assumptions, the DSAM can segment the network in several places to help direct technicians to specific 'legs' of the network when problems arise.

Typically a VoIP install will use DOCSIS techniques to ensure that the voice packets are treated with a higher priority than the data packets that are received from the same cable modem. This technique is referred to as Quality of Service (QoS). Each of the above tests has the ability to be run with or without VoIP QoS. Refer to Appendix A: for details on how to enable VoIP QoS for the VoIPCheck test.

3.1.3 CMTS Loop:

Upon entering the VoIPCheck test, the DSAM will enter the CMTS Loop Screen (Figure 1). This screen provides packet information, such as Packetloss, Delay, and jitter, as well as Quality information such as a Mean Opinion Score (MOS) and R-Value. The test is continuously sending and receiving VoIP packets to and from the CMTS during the test, and the packet statistics and quality metrics are updated approximately every 2.5 seconds.

The VoIPCheck test treats every 1.25 seconds as a new 'call', so that the quality measurements (MOS and R-Value) are susceptible to issues even after the test has been running for long periods of time. If this were not the case, quality measurements would all trend toward the average, and become much less instantaneous as the test continued to run. By handling the measurement in this manner, the DSAM can show you instantaneous (Current) quality, average (Avg) quality, and minimum (Min) quality of the drop since the test was initiated.

CMTS Loop		CODEC: G.711u Jitter Buffer Size: 40ms		
Packets		Current	Max	Avg
PacketLoss		0.0%	0.0%	0.0%
Jitter		2.6	19.0	2.5
Delay		12.4	22.1	12.3
Quality		Current	Min	Avg
MOS		4.2	4.2	4.2
R-value		93	93	93

Figure 1:
 CMTS Loop Screen
 Displays the results between the CMTS and the DSAM meter.

3.1.4 Segmentation:

In addition to the CMTS Loop display, the VoIPCheck measurement also provides a segmentation view of the system. This view of the measurement allows technicians to determine which ‘legs’ of the network are causing issues with the quality of the voice service. The test is performed exactly like the CMTS Loop test, and it alternates between testing between the CMTS and the Advanced IP Server. Performing the test in this manner allows the DSAM to provide measurements that are specific to individual ‘legs’ of the path being tested. This test can be used to determine if issues with voice services reside within the HFC (CMTS Loop) or the IP Backbone (Server Loop). Varying views of the results allow packet statistics (Figure 2) or quality values (Figure 3) to be displayed when desired. The ‘Data Menu’ can be used to select which data results the technician wants to display (Current, Average, Extreme ...).

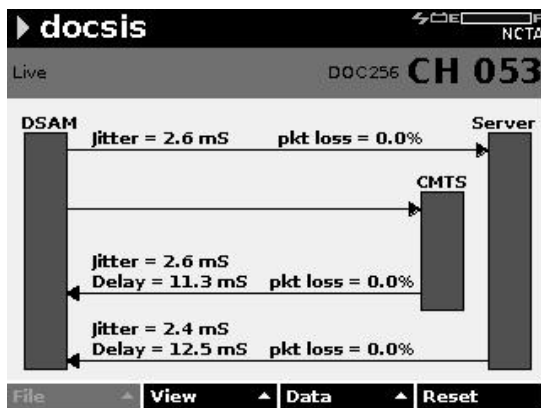


Figure 2:
 Segmentation Packet Stats screen

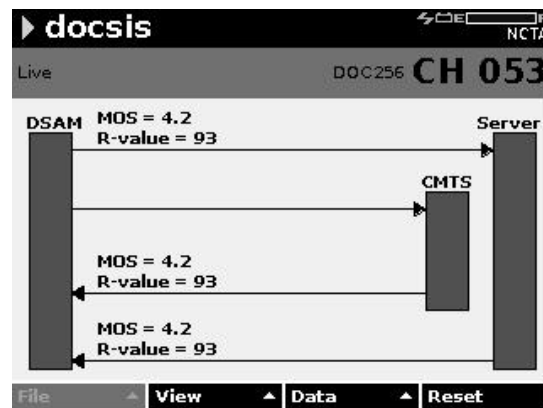


Figure 3:
 Segmentation Call Quality screen

3.1.5 Configuration:

To successfully perform the server loop during segmentation, the IP address and port number of the server must be configured within the DSAM, as shown in Figure 4. Additionally, the user has the ability to select which CODEC will be simulated by the test, and the size of the jitter buffer to simulate. The CODEC selection affects the values that will be calculated in the MOS and R-Value. Due to compression and bandwidth requirements, CODEC’s will have different theoretical max values for MOS and R-Value. (i.e. the maximum value for MOS on the G.711 protocol is 4.2 instead of the 5.0 that may be expected) The port number that is entered for the server must match the port number

that is configured on the server. Additionally, the port number that is entered here will be utilized when setting up QoS for the VoIPCheck test, as will be described in Appendix A.

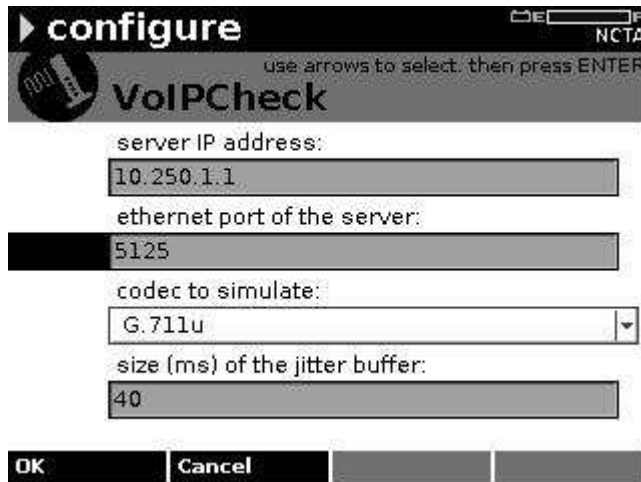


Figure 4:
VoIPCheck Configuration screen
The Advanced IP Server information can be configured as well as the CODEC used and jitter buffer size.

NOTE: Port numbers 5100 and 5101 are reserved for the current data packet loss and throughput measurements and should be avoided as the VoIPCheck port.

3.1.6 Summary

The DSAM VoIPCheck test is invaluable in two major areas of CATV maintenance. First, drop qualification becomes straight forward at the install. One test will tell the installer if the current setup will support the necessary packet flow to provide high quality voice service. Second, when problems do arise, the VoIPCheck test can be used by technicians to segment the network and determine the origin of the issue. Thus reducing time required to find and fix voice quality issues.

3.2 PacketCable™ VoIP:

3.2.1 Overview:

The DSAM PacketCable™ VoIP measurement is intended to use the eMTA within the DSAM to perform full PacketCable™ VoIP provisioning, signaling, and voice quality testing. The DSAM contains an eMTA and a speakerphone functionality that has the ability to join a PacketCable™ network and function as a telephone. Phone calls can be initiated or received by the DSAM, and quality of the call can be quickly assessed.

3.2.2 MTA Provisioning:

The PacketCable™ VoIP test begins by provisioning the CM within the DSAM. Following this, it begins the process of provisioning the eMTA. Each step along the way is displayed to the technician, and should any step fail, a dialog will display describing the error information. Once online, the eMTA continues by initializing endpoint(s) with the Call Management System (CMS), and actually obtains a phone number(s). Once all provisioning is complete (Figure 5) the DSAM will be 'ready' to place and receive phone calls (Figure 6).

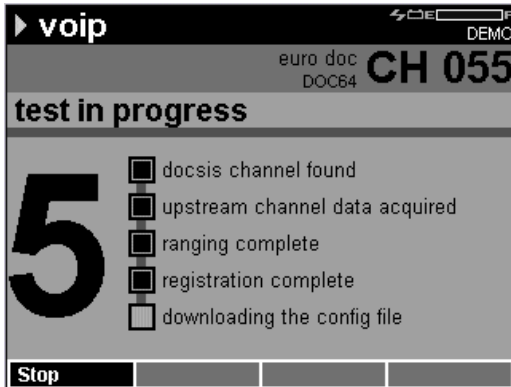


Figure 5:
 PacketCable™ VoIP provisioning and signaling steps screen

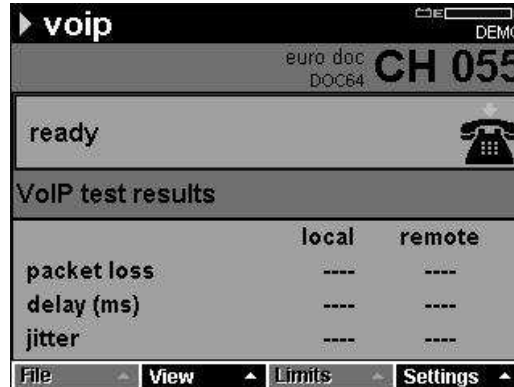


Figure 6:
 PacketCable™ VoIP 'Ready' screen

NOTE: After the DSAM joins the network the technician can initiate or receive phone calls using the DSAM user interface.

3.2.3 Signaling:

PacketCable uses NCS signaling to initiate phone calls. The DSAM is capability of exercising these signaling capabilities. The DSAM can be utilized to test for dial tone or make phone calls to test call quality. Through its ability to make and receive phone calls, the DSAM is proving the signaling paths to all of the necessary servers in the system. When properly provisioned within a PacketCable System, the DSAM has the ability to test signaling to both on-net and off-net destinations. The only difference between the two, from the DSAM user's perspective, is the phone number that the technician enters to test.

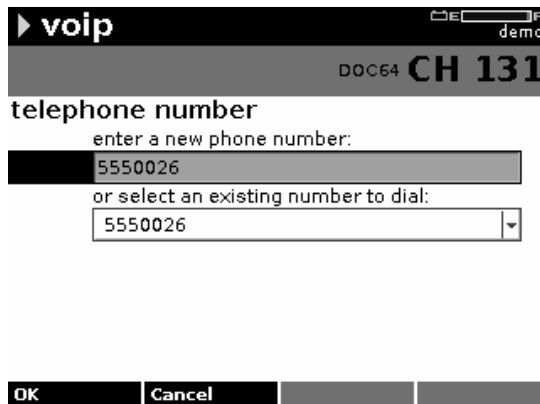


Figure 7:
 PacketCable™ VoIP telephone number screen. The user can enter any telephone number either on or off the network.

During the call setup and teardown portion of the signaling, the DSAM will perform any Dynamic Quality of Service (DQoS) setup that is necessary based on the PacketCable™ network configuration. This is the same setup that would be required by any MTA on the system. Given this, the call quality statistics that the DSAM will be measuring are based on the actual setup that an MTA would be experiencing in a similar phone call.

3.2.4 Voice Quality Testing:

Once a call has been initiated or received by the DSAM, it will enter the ‘Ongoing Call’ screen. In this mode, the DSAM is measuring packet statistics, as well as continuously recording the quality of the call. Providing Packetloss, Jitter, and Delay measurements for both the local and remote endpoints, gives the technician a picture of the upstream and downstream packet performance. Additionally, the DSAM’s ability to determine a MOS score will allow technicians the ability to quickly assess the call quality that a customer would experience on the drop at this location. If the quality is determined to be unsatisfactory, the technician can analyze the packet statistics to determine whether the low quality is a product of packet loss, delay, or jitter, and can then take the appropriate action to fix the problem.

		Current	Max	Avg
Local	packet loss	3.8 %	8.0 %	4.0 %
	jitter	5	5	2
	delay (ms)	45	45	12
Remote	packet loss	4.0 %	4.0 %	3.7 %
	jitter	4	7	3
	delay (ms)	45	45	12

Figure 8:
 PacketCable™ VoIP ‘Ongoing Call’ packet statistics screen

	Current	Min	Avg
R-Value Listening	79	78	79
R-Value Conversational	78	77	78
MOS Listening	3.8	3.8	3.8
MOS Conversational	3.8	3.7	3.8

Figure 9:
 PacketCable™ VoIP ‘Ongoing Call’ quality screen

3.2.5 Summary

The DSAM PacketCable VoIP test provides the best overall method for maintaining a PacketCable™ VoIP network. The ability to verify provisioning and CMS servers are all operational is a crucial part of MTA installation and troubleshooting. The ability to quantify the quality of the connected call is the next required functionality. An additional benefit of having test equipment that can join the network and participate as an MTA, is the ability to exactly duplicate a customer complaint. When customers are having issues with calls to a specific number, the DSAM PacketCable VoIP option is able to actually call the same number, and quantify the problem. This is a major step forward in VoIP testing.

4 SUMMARY:

JDSU has worked to provide the most comprehensive VoIP solution possible to the CATV marketplace. Utilizing the VoIPCheck, and PacketCable™ VoIP options within the DSAM platform, during installation, maintenance, and troubleshooting of the network will provide cable operators with the best overall handheld VoIP solution on the market.

5 Appendix A: Setting up QoS for VoIPCheck

5.1 Description:

To make the VoIPCheck offering in DSAM more closely resemble the quality that is being attained on the actual VoIP network, it is necessary to employ the same QoS within the two environments. With this said, this section will explain how to create a CM configuration file for the DSAM that will force the VoIPCheck to utilize a Unsolicited Grant Service (UGS) flow on the upstream, and a guaranteed bandwidth flow on the downstream.

5.2 Setting up the flows:

To enable QoS on the VoIPCheck service, the following upstream and downstream flows must first be included in the DSAM CM config file.

UsServiceFlow

```
{
  UsServiceFlowRef 1;
  QoSParamSetType 7;
  SchedulingType 5; ---This is UGS
  RequestOrTxPolicy 0x0000017f;
  UnsolicitedGrantSize 240; ---This is the value for g.711
  NominalGrantInterval 20000; ---This is the value for g.711
  ToleratedGrantJitter 5000;
  GrantsPerInterval 1;
}
```

DsServiceFlow

```
{
  DsServiceFlowRef 4;
  QoSParamSetType 7;
  TrafficPriority 7;
  MaxRateSustained 512000;
  MaxTrafficBurst 375000;
  MinReservedRate 512000;
}
```

5.3 Including the classifiers:

Simply including the flows will not force the VoIPCheck test to use the appropriate flows. The following classifiers must be included in the same configuration file to tell the CM to force the traffic onto the appropriate flows.

NOTE: References to “VoIPCheck port” below, refer to the port that is configured for the server of the VoIPCheck test described in section 3.1.5.

NOTE: Port numbers 5100 and 5101 are reserved for the current data packet loss and throughput measurements and should be avoided as the VoIPCheck port.

```
UsPacketClass
{
    ClassifierRef 1;
    ServiceFlowRef 1;
    RulePriority 1;
    ActivationState 1;
    IpPacketClassifier
    {
        IpProto 17;
        DstPortStart "VoIPCheck port";
        DstPortEnd "VoIPCheck port";
    }
}

DsPacketClass
{
    ClassifierRef 2;
    ServiceFlowRef 4;    ---from flows defined above
    RulePriority 1;
    ActivationState 1;
    IpPacketClassifier
    {
        IpProto 17;
        DstPortStart "VoIPCheck port";
        DstPortEnd "VoIPCheck port";
    }
}
```

5.4 Summary:

Including the before mentioned entries into the CM configuration file that is delivered will allow the VoIPCheck to utilize a UGS flow of all upstream packets, and a guaranteed bandwidth flow for all downstream packets. With this setup, VoIPCheck quality measurements will parallel the actual VoIP traffic on the system very closely. This will also allow the current Packetloss and throughput measurements to operate in the current mode. They will not use either of the flows that are defined above, but rather will continue to use the best-effort flows as would a CPE device behind a CM.