Microsoft TV Test

by John Williams

Microsoft has developed a TV middleware platform called Mediaroom that focuses upon Internet Protocol (IP) video services offering significant enhancements in delivered quality compared to networks that do not use their middleware. These enhancements include content control through embedded digital rights management (DRM) and encrypted content, reduced channel change times, and robust error recovery mechanisms. JDSU has developed specific test support for service providers using Mediaroom-based networks. While Mediaroom has now been extended to work with and through other devices, including Windows computers, compatible smart phones, and the Xbox 360, this application note deals only with IPTV service delivered to typical residential subscribers using set top boxes (STBs) and a TV display device.

Technology Background

The JDSU Microsoft TV (MSTV) Test Suite analyzes two significant portions of Mediaroom: instant channel change (ICC) and Reliable User Datagram Protocol (R-UDP), the error recovery mechanism. Both of these elements involve a software client in a STB that can communicate with Distribution Servers (D-Servers) located in the network, typically in video serving offices at the network edge.

In order to analyze the unicast flows that make up the ICC and R-UDP elements of Mediaroom, the tester must be in a Monitor mode so that it can "observe" the unicast flows as well as the associated signaling. However, the tester can work in a Terminate mode and join multicast video programming in a Mediaroom network conducting the normal broadcast IP video analysis.

ICC Technology

While the Microsoft system uses the basic Internet Group Management Protocol (IGMP) leave/join protocol to change channels, an additional operation has been added so that a D-Server specifically paired with an STB client becomes part of the process and can burst a unicast stream to the STB carrying a segment of the target channel's data flow, thus filling the STB decode buffer more quickly. The STB client and the format of the multicast stream control this, however, the acquisition servers (A-Servers) modify the stream in the headend adding, for example, random access points (RAPs) to facilitate the splicing of the unicast and multicast streams. This unicast burst allows play-out of the decoded video program to the display ahead of the multicast joined stream's arrival, thus shortening the channel change time.

In both the ICC and the R-UDP cases, D-Servers are added to the network, typically located near the edge of the network in video serving offices. Each D-Server is assigned a specific set of STBs by IP address. The protocol messaging flows between the STB and an assigned D-server are outlined. In Figure 1, the STB initiates the IGMP join message. At the same time, it initiates the ICC protocol messaging. A rotating buffer holds a copy of the live stream's last few seconds of content in the D-server. When requested, it will send a unicast burst to the requesting STB, typically at 130 percent of the normal flow rate.

The diagram below outlines the data flows involved for ICC operation.

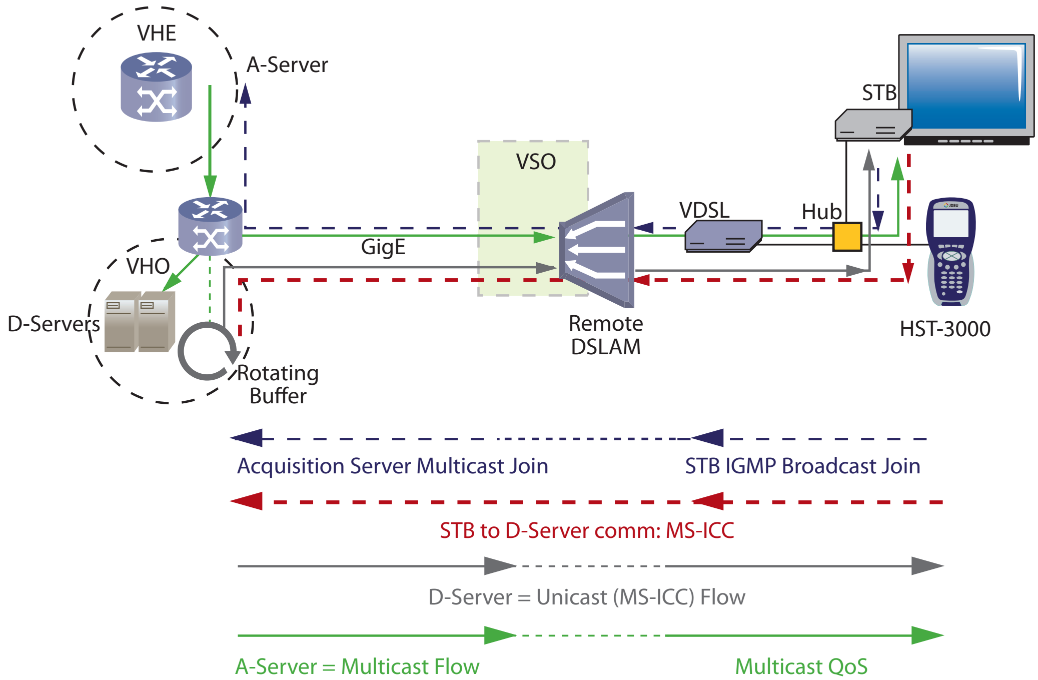


Figure 1. STP initiating the IGMP join message as well as the ICC Protocol message

R-UDP Technology

The error recovery mechanism called R-UDP uses the same D-Server and rotating buffer. The STB client uses a special R-UDP protocol to request retransmission of specific lost Real Time Protocol (RTP), RFC 2250, packets. In this operation, the specific RTP packet sequence numbers for lost packet events are reported to the D-server, which then sends the requested packets to the STB where the MS client splices the unicast packet flow from the D-Server into the multicast program flow. The buffers in the STBs hold approximately 1 second of stream video, giving the client enough time to complete the splice operation without impact to the play-out flow to the display.

Figure 2 outlines the data flows involved for R-UDP operation.

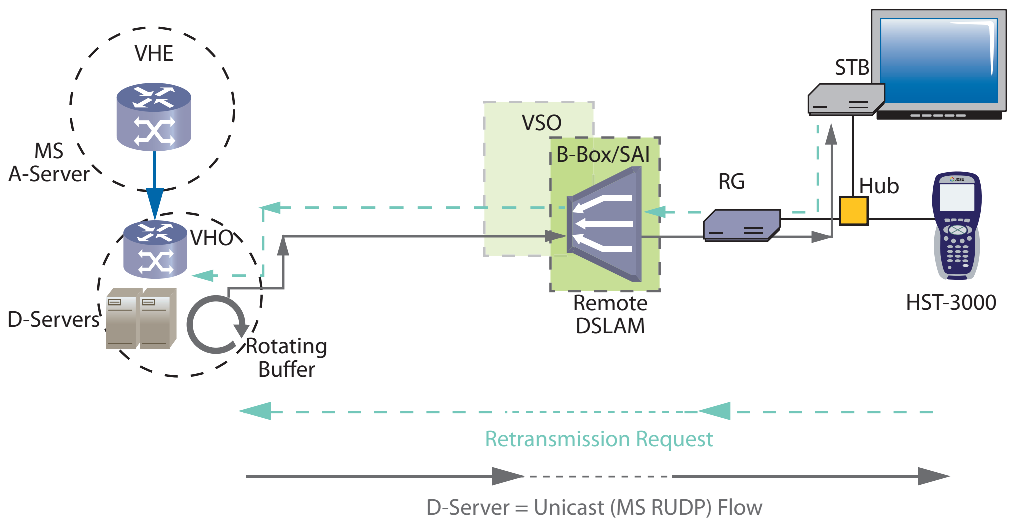


Figure 2. Data flows involved in R-UDP

Of course, all of these operations have limitations. The MSTV test suite first validates proper operation using the simple quality of service (QoS) analysis summary for key metrics. Upon meeting all of the thresholds, the technician can easily save the results and move on. However, when necessary, it is possible to troubleshoot a failure using in-depth analysis.

Test Analysis

Test Access

The HST-3000 supports a selectable Through mode using the HST inline with the installed xDSL SIM interface and the Ethernet 10/100 I/F. It also supports a Monitor mode where the HST uses the base Ethernet I/F to gain access to a data flow that requires an external hub. A small USB-powered external hub is generally available on the market that provides this simple Ethernet access.

The HST screen in Figure 3 shows the monitor screen with two active streams present that are automatically analyzed once a stream is up.

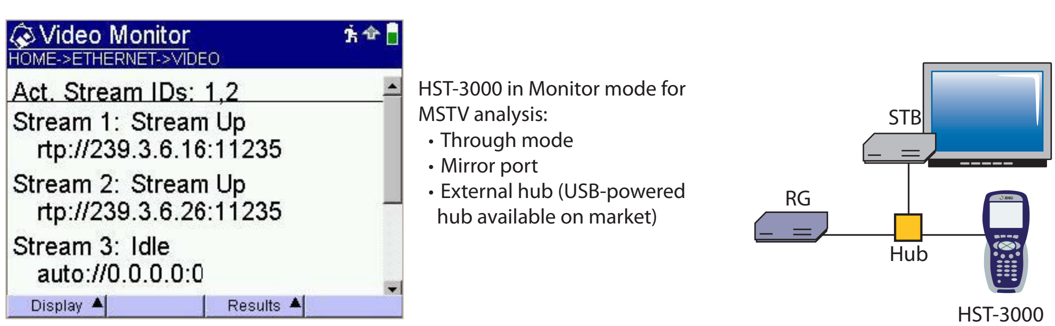


Figure 3. HST Video Monitor screen with active streams present

Test Analysis—Turn-up QoS

The HST-3000 provides full packet analysis of multicast streams. To view the analysis for Stream 1, press the Right arrow NAV key to reveal the QoS screen. The QoS analysis screen includes the critical metrics, instantly providing a view of the stream's health. The Pass and Fail thresholds can be set or tuned to the specific network design in the configuration menus.

PCR jitter the critical timing reference signal with typical thresholds of 40 to 50 ms.

R-UDP refers to the MSTV packet loss mechanism. Uncorrected % deals with packet loss that remains uncorrected and, therefore, will likely impact the quality of experience (QoE). This metric replaces the Continuity Error metric in a non-Mediaroom network.

Error Indicator as set by the encoder signifies a problem with the input or source data.

Latency refers to the IGMP join latency, which is typically the longer step in the channel change sequence in the IGMP protocol. Leave latency indicates the elapsed time from when the message was sent to the cessation of the stream. A zero Leave Latency indicates that no packets were received after the leave message was sent. The total of the two is typically referred to as the Zap time.

For more message flow detail, see Figure 4 at the end of this document which provides a diagram of the Mediaroom Message Flow.

If the overall QoS state reads Pass, the turn-up test is complete. The QoS of the stream is good and performance is within established levels. Users can press the Up and Down Navigation keys to move between streams for each screen selected.

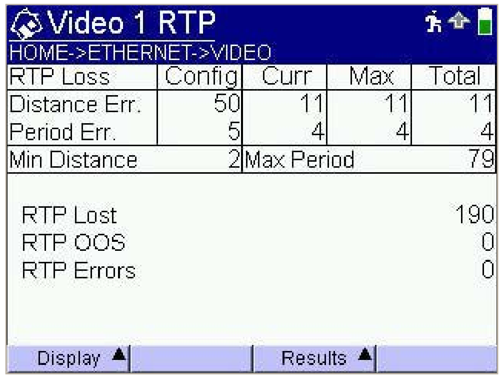
If the overall status reads Fail, users can press the Navigation keys Right or Left to move through the detailed analysis screens for troubleshooting. Users can press the Results soft key and then select Save Results to save all of the detailed analysis screen data so that, as in many processes, another technician may complete the troubleshooting process.

Overall QoS State screen

Test Analysis—Troubleshooting

Packet Analysis

While many root-cause issues may be at work in any given situation, packet loss is typically the most critical area. Users can press the Right Navigation key to get to the RTP Packet Analysis screen.

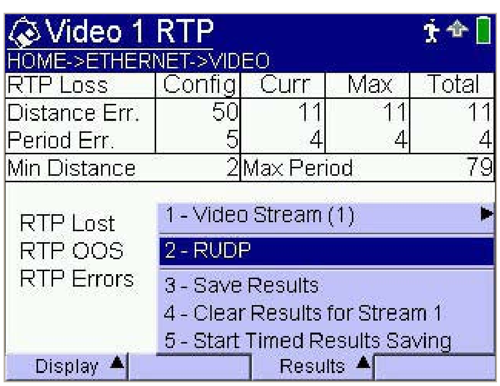


RTP Packet analysis

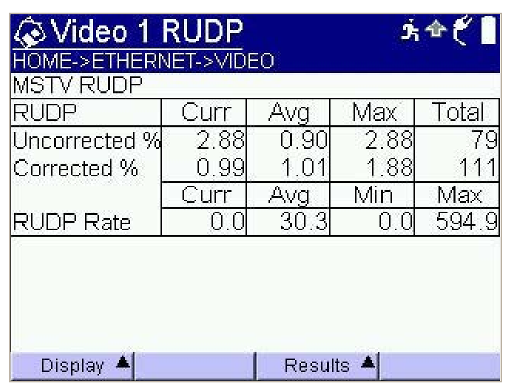
RTP packet analysis follows RFC 3357 in which a Period, when packet loss occurs (defined as the number of RTP packets lost), and the Distance (defined as the number of properly received packets) between loss events, enables detailed analysis for Mediaroom RTP streams. Users can analyze any error recovery mechanism's performance using these concepts. In the Mediaroom network, the rotating recovery data buffer is time based, as mentioned previously. Thus, establishing thresholds for performance limits is required for each stream rate used. The Max Period threshold is set at 14 times the stream rate in Mbps. The Min Distance threshold is set at 3 times the Max Period. The minimum spacing refers to having too many events that occur in a short time period. Mediaroom evaluates these conditions and in some cases lumps a group of small loss events together in the unicast burst in the hopes that the client can then determine which packets to keep and which to ignore.

The out-of-sequence (OOS) packet count is important, because it indicates occurrences that are only possible deep within the network where congestion typically occurs.

These two screens show the navigation steps used to perform a detailed analysis of the R-UDP error recovery actions that may have occurred. Selecting R-UDP under the Results key moves the user to the R-UDP screen that provides the details of the function. If it shows any uncorrected activity, one can conclude that errors reached the display and impacted QoE.

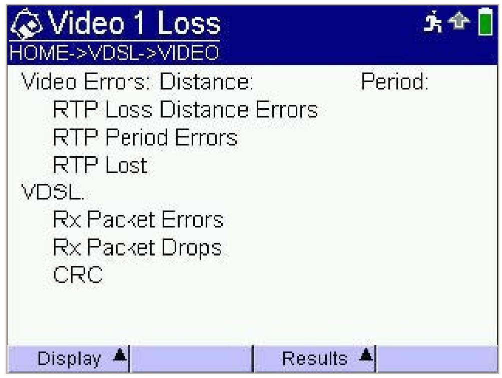


R-UDP navigation



R-UDP analysis

The next analysis screen combines the correlation between RTP packet analysis and the physical layer of the test access point.

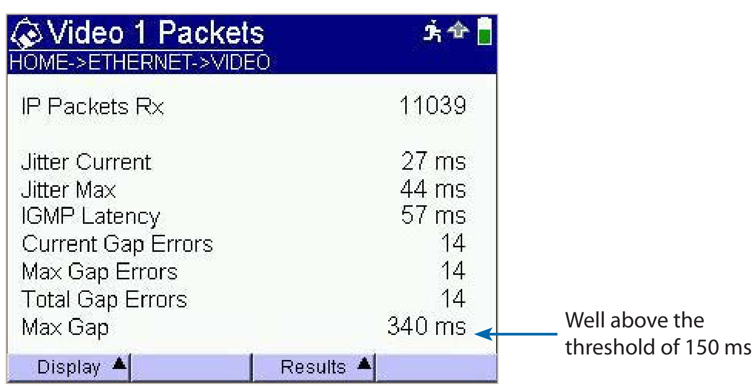


RTP loss correlation with test interface media layer

This example screen of RTP packet analysis using Distance and Period enables easy comparison with the VDSL stats. If it indicated any RTP loss, the correlation to any CRC (cyclic redundancy check) errors would be key. With no CRC errors on the VDSL (very high speed digital subscriber line) interface, packet loss is coming from deeper within the network, northbound of the DSLAM (Digital Subscriber Line Access Multiplexer). If CRC errors are present, then the copper loop carrying the VDSL service is either partially or fully responsible for the RTP loss. This analysis enables users to quickly accomplish root-cause sectionalization.

Stream Analysis

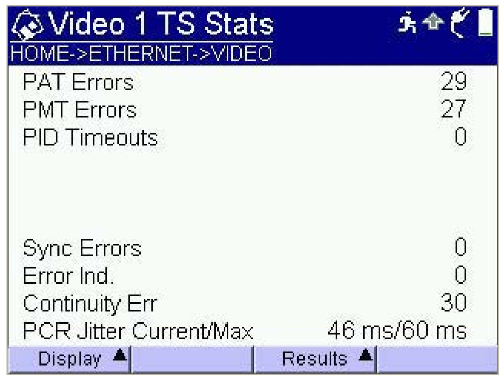
The Video 1 Packets screen summarizes several metrics for easy comparison. More importantly, it adds a time-based gap analysis that provides critical insight into the rotating buffers in the D-Servers, which are also time based.



IP Packet flow details

A gap is defined as a hole in time. Errors are based on a configurable threshold, typically 150 ms. A gap may result from excessive jitter, a program insertion gap, or packet loss.

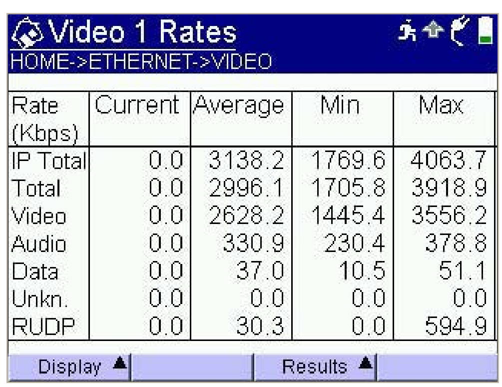
The Video 1 TS Stats screen provides additional detail regarding the video MPEG-2 (Motion Picture Experts Group 2) Transport Stream packet flow. The lower half of the screen summarizes several QoS metrics for easy comparison with the status of the Program Specific Information (PSI) represented by the table data contained in the flow and the presence of that data on a regular basis or repetition .



MPEG-2 Transport Stream packet flow details

Program Association Table (PAT) and Program Management Table (PMT) data are inserted in the packet flows enabling the decoder to organize the multiple packet flows, video packets, and audio packets that make up the program content for decoding and presentation. It shows errors related to conditions such as a scrambled mode being on and PID time-outs related to the presence of the data in the stream based on a threshold setting. In a typical non-Mediaroom network, the PSI data must be present every 500 ms; however, a range of 1.5 to 2 seconds is typical in an MSTV network.

Detailed analysis is provided for the data rates associated with each portion of the multimedia program stream. The minimum (Min) and maximum (Max) data rate history is very useful, because the Min for the video level indicates a program flow interruption which would impact QoE. The Max total shows the maximum data rate reached during the test period which is useful when analyzing a variable bit rate (VBR) stream. It also clearly shows any R-UDP traffic that may have occurred during the test period.


Video 1 Rates

HOME->ETHERNET->VIDEO

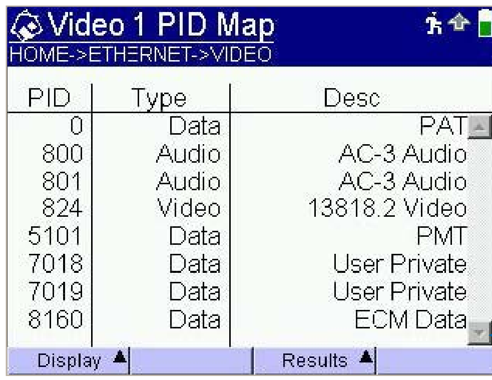
Rate (Kbps)	Current	Average	Min	Max
IP Total	0.0	3138.2	1769.6	4063.7
Total	0.0	2996.1	1705.8	3918.9
Video	0.0	2628.2	1445.4	3556.2
Audio	0.0	330.9	230.4	378.8
Data	0.0	37.0	10.5	51.1
Unkn.	0.0	0.0	0.0	0.0
RUDP	0.0	30.3	0.0	594.9

Display Results

Bandwidth details

Unknown data, in this example, is typically Pad data added to a stream to keep a constant bit rate for a CBR (constant bit rate) stream.

The Packet Identification (PID) Map screen provides a detailed view of all of the program pieces since it identifies each individual packet flow by the PID number. Showing more than one audio PID could indicate more than one language, but the details are specific for each network design. The Description field adds content info. Video 13818 is an MPEG-2 compression scheme. Other descriptions may include 14496 or H.24 which is MPEG-4 and VC-1 which is the SMPTE name for a MSTV compression technology. Private Data may be audio in some implementations. ECM (Entitlement Control Management) is the flow carrying DRM and encryption data to the client in the STB.



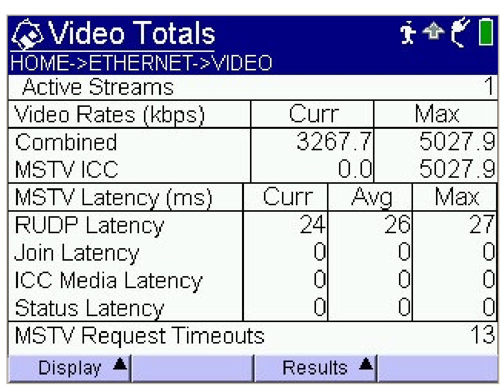
The scroll bar indicates that more PID data is available in this example.

PID Map

Latency Analysis

The next screen provides the details of the operations of the ICC mechanisms. Stream analysis provides a combined multicast bandwidth (BW) and a combined ICC flow BW during the test period.

- **R-UDP latency**: the elapsed time (in ms) from request to receipt of the first retransmitted media packet.
- **ICC Media Latency**: the elapsed time (in ms) from the join request to receipt of the first ICC media packet.
- **Status Latency**: the elapsed time (in ms) from the last Status Request to the last Status Response. This communication occurs approximately every 5 seconds.
- **Join Latency**: the elapsed time (in ms) from the join message Tx to the Rx of the acknowledgment .



Latency and ICC details

Analysis Logs

The **Monitor log** included as a standard part of the IP Video test suite has been expanded to include a decoded presentation of the new Mediaroom protocol messages that include:

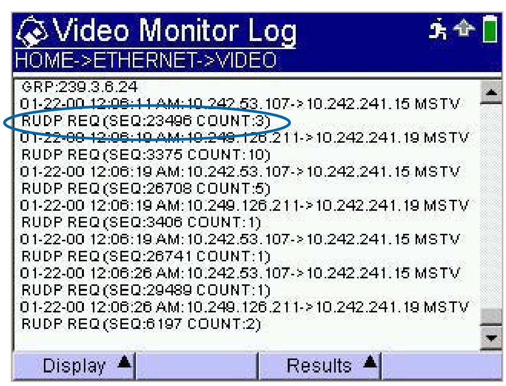
- Join Request
- Retry (R-UDP) Request (including the first sequence number lost and the hole size).

  Note: The absence of 1 to 1 correspondence between loss events and "Retry Requests", because holes within 100 ms are typically grouped as one.
- Leave
- Burst Complete Acknowledgment
- Join Response (with Burst ON or OFF)
- Burst Complete
- Known Hole in Stream (currently not implemented by Microsoft)

Description	Sent By
Join Request	Client
Retry Request	Client
Leave	Client
Status (AKA Stat or Ping)	Client
Ack Burst Complete	Client
Join Response	D-Server
Burst Complete	D-Server
Status Response	D-Server

Mediaroom Messages

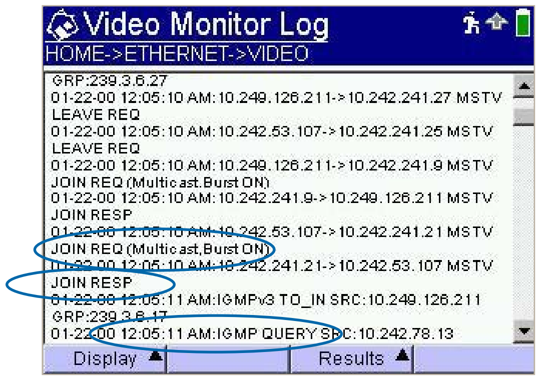
This example of a Monitor Log screen shows a typical sequence of entries, in this case R-UDP-related messaging. The log shows the time stamp for each entry, the IP address of the sender followed by the destination IP address, and the message content summary.



Typical sequence of entries

In the example highlighted, the STB requests that three lost packets be sent to repair a hole with the first RTP sequence number of the lost packets as 23496. It also shows other R-UDP messages.

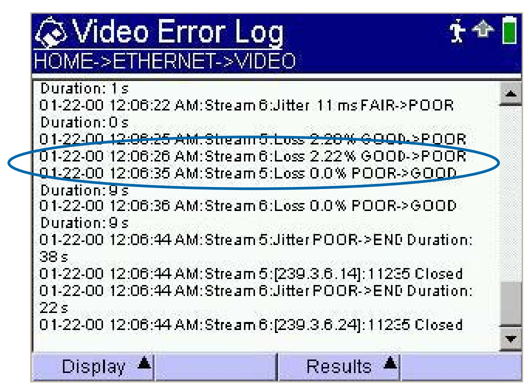
This Monitor Log screen shows details of the IGMP messaging. Users can analyze in detail Leave and Join messages, including the join response from the D-server. The Join message includes the request to have the ICC unicast burst occur: Burst ON.



Monitor Log details

The Monitor Log screen shows details of the IGMP Group Query protocol. All STBs respond to confirm that they want to remain in a particular group, meaning it will provide access to that program material at the STB. If the D-Server does not get its required response, the program flow is terminated for the IP address of that particular STB.

The **Error Log** includes only entries defined as events related to the test suite and not protocol errors. It logs all threshold crossings from Pass to Fail or Fail to Pass along with the elapsed time in the failed state.



Error Log details

Thresholds

Various thresholds must be configured to conduct the MSTV analysis. However, a specific threshold must be tuned to a given network. The thresholds listed in the MSTV column in Table 1 indicate JDSU default settings and serve as a starting point for refinement. The column for "multicast" thresholds below apply to testing in a Mediaroom network in a terminate mode where the test set is joining a multicast stream. In this terminate mode the unicast flow analysis is not possible, but stream analysis does provide network performance detail.

Metric	MSTV	Multicast
ICC Latency	40 ms	NA
R-UDP Latency	500 ms	NA
Status Latency	40 ms	NA
PID Timeout	1000 ms	1000 ms
PAT/PMT Error	1500 – 2000 ms (Adjust to A-Server settings)	1500 – 2000 ms in an MSTV network; 500 ms in Standard networks
Gap Error	150 ms (Adjust to D-Server settings)	150 ms (Adjust to D-Server settings), NA for Standard networks
Max Period	14 x M of Max stream rate	Same if in an MSTV network. Set Threshold to match any FEC settings if not an MSTV network; if none, set to 5
Min Distance	14 x M x 3 (Adjust to D-Server settings)	Same if in an MSTV network. Set Threshold to match any FEC settings if not in an MSTV; if none, set to 50
PCR Jitter	50 ms, 100 ms Fail (Adjust to network design)	Same if in MSTV, 100 ms Pass; 120 ms Fail if another network, or tune to network
IGMP latency (Join latency)	40 ms, 50 ms Fail	Same if in MSTV; 500 ms if not
Loss (Un-corrected R-UDP in MSTV; Continuity Error in non-MSTV)	0.01% pass, 0.01% Fail (After R-UDP tried to fix packet loss)	0.01% Pass, 0.2% Fail
Error Indicator	1	1

Table 1. Metrics and Thresholds

Note: The minimum distance threshold setting in the MSTV column is one that may significantly change, because R-UDP operations sometimes group several small error events together.

Message Flows

Figure 4 outlines the message flows between the Mediaroom client in the STB and the D-Server which is helpful in analyzing the Monitor Log data.

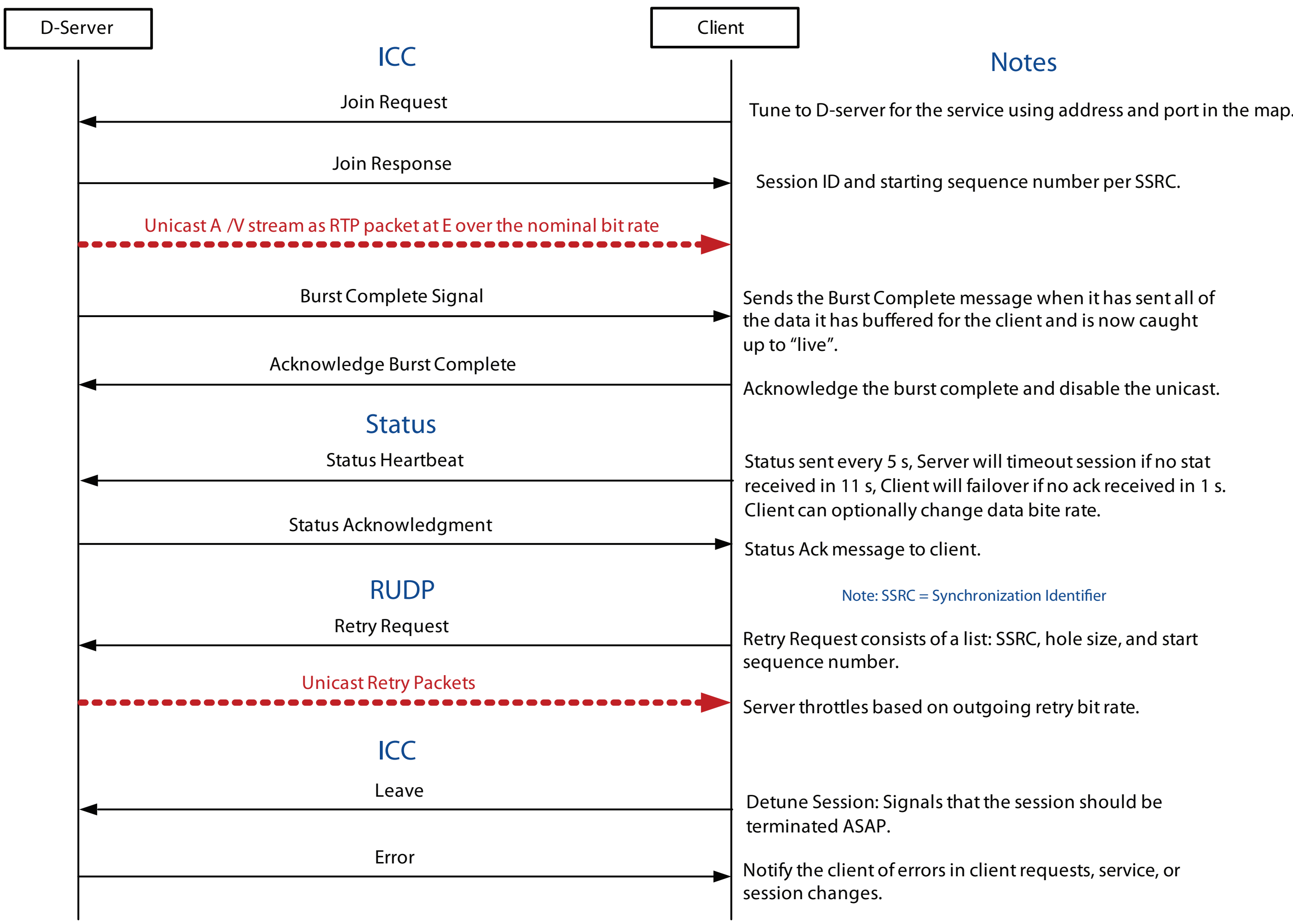


Figure 4. Message flows between the Mediaroom client and the STB

Summary

In summary, the MSTV option adds to the existing IP Video option to provide analysis of the special Mediaroom unicast flows and associated messaging. While all of this is very complex, turn-up is simple. The QoS screen tells the story: if Pass then all is well; if Fail, in--depth trouble analysis is supported. The detailed troubleshooting requires more knowledge, but root-cause analysis is possible with the one tool as necessary.

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