HD Radio™ SFN Broadcast Test Plan & Results

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TRADEMARKS

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1 Scope

1.1 System Overview

The iBiquity Digital Corporation HD Radio™ system is designed to permit a smooth evolution from current analog Amplitude Modulation (AM) and Frequency Modulation (FM) radio to a fully digital In-Band On-Channel (IBOC) system. This system delivers digital audio and data services to mobile, portable, and fixed receivers from terrestrial transmitters in the existing Medium Frequency (MF) and Very High Frequency (VHF) radio bands. Broadcasters may continue to transmit analog AM and FM simultaneously with the new, higher-quality and more robust digital signals, allowing themselves and their listeners to convert from analog to digital radio while maintaining their current frequency allocations.

1.2 Document Overview

This document describes test setup, test equipment, and test description used for verifying the broadcast system for use with single frequency networks (SFN). The document also presents results and observations of the tests performed.

2 Referenced Documents

[1] iBiquity Digital Corporation, "HD Radio Single Frequency Network Design Description", Document Number: SY_IDD_1011

[2] iBiquity Digital Corporation, "HD Radio Single Frequency Software Design Description", Document Number: SY_SDD_2626

[3] B. Kroeger, "Guidelines for FM HD Radio Repeaters", iBiquity Digital internal document, July 3rd 2003

[4] iBiquity Digital Corporation, "HD Radio™ Commercial Exciter Self Certification Plan ,"
Document Number: T1A-CERT-1070

3 Abbreviations and Conventions

3.1 Abbreviations and Acronyms

AAS	Advanced Application Services
AM	Amplitude Modulation
AES	Audio Engineering Society
CD	Compact Disk
EASU	Exciter Auxiliary Service Unit
FM	Frequency Modulation
GPS	Global Positioning System
HDC	HD Codec
HDP	HD Protocol
IBOC	In-Band On-Channel
IF	Intermediate Frequency
IP	Internet Protocol
MAM	Market Area Monitor
MPS	Main Program Service
MF	Medium Frequency
PDU	Protocol Data Unit
PPS	Pulse Per Second
PSD	Program Service Data
RF	Radio Frequency
SFN	Single Frequency Network
SPS	Secondary Program Service
STL	Studio Transmitter Link
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
VHF	Very High Frequency

3.2 Presentation Conventions

Unless otherwise noted, the following conventions apply to this document:

- Information enclosed in braces { } is either unavailable at the present time or subject to change.
- Glossary terms are presented in italics upon their first usage in the text.
- All vectors are indexed starting with 0.
- The element of a vector with the lowest index is considered to be first.
- In drawings and tables, the leftmost bit is considered to occur first.
- Bit 0 of a byte or word is considered the least significant bit.
- In representations of binary numbers, the least significant bit is on the right.
- When presenting the dimensions of a matrix, the number of rows is given first (e.g., an n x m matrix has n rows and m columns).
- In timing diagrams, earliest time is on the left.

4 Test Goals and Setup Descriptions

4.1 Goals

The goals of these laboratory tests are: (1) verify the broadcast system components can be aligned to the necessary resolution, under both synchronous and asynchronous start conditions. (2) Verify the delay of the individual nodes can be accurately adjusted. (3) Verify the broadcast system component alignments are robust to network delays and bandwidth restrictions and determine the range of tolerances to these distortions. (4) Verify, to the extent possible, digital receiver performance as a function of signal level and relative delay.

4.2 Equipment

The key components of the SFN test bed are shown in Figure 4-1. A CD player serves as the source for the audio. The AES audio is fed into an EASU where it is rate converted into a common word clock derived from a GPS time base. The rate converted audio and a 10 MHz frequency reference are fed into the Exporter. The audio is delayed and output to an FM modulator to serve as the common analog host[1]. The Exporter processes the audio and outputs Layer 1 PDUs to each of the two Exgines via an Ethernet based E2X link. A network shaper is inserted into the path of one of the Exgines to simulate network delays, bandwidth restrictions, and other real world STL conditions. Each Exgine is supplied its 10 MHz reference by independent EASUs locked to GPS. The RF from the Exgines and the FM modulator are combined and fed into both a Linux receiver and a IDM 352-based Market Area Monitor (MAM)[2]. One of the RF paths is first fed into a variable attenuator before being combined with the other two signals. The output from the Linux receiver or MAM can be fed into the oscilloscope to verify alignment with the analog signal. In addition, the MAM can perform split mode correlations between the analog and digital channels, using the analog as a reference. This would show either the sum or separate digital signals aligned in audio time to the analog signal. The IF output of the Exgines are fed into a dual channel simultaneous sample and hold oscilloscope for analysis. Finally, the RF from a single Exgine is fed into a time locked spectrum analyzer to verify frequency accuracy.

The slip buffers in the Exgines are used to investigate receiver performance when the two nodes are delayed relative to each other. However, the resolution on the Exgine delay is only ~1.3 µsec (i.e. one FM baseband sample), whereas the multipath interference pattern would typically repeat every 10 nsec. If the receiver performance needs to be studied at these delay resolutions then either an external RF delay unit must be used or a software fractional delay must be implemented in the Exgine. A software fractional delay process could be problematic due to the high processing requirements and may not fit in the current Exgine host processing bounds. In any event, high resolution delay performance is not addressed in these lab tests.

[1] FM analog SFN performance is not addresses in these lab tests.
[2] A MAM is a baseband receiver with a PC as a host processor

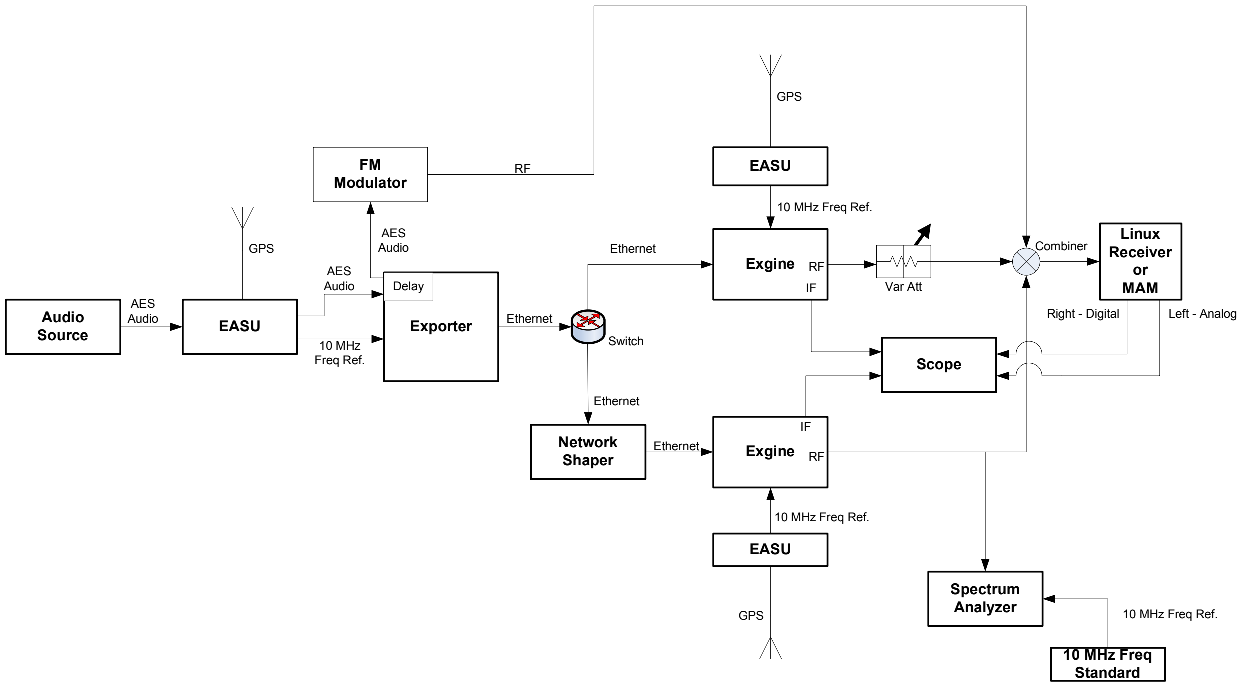

Figure 4-1 SFN Test bed

4.3 Software

The software used for these tests is an engineering or prerelease version of the MPS Framework v4.3.x. The software can be obtained from the various broadcast equipment manufactures or alternatively, the software can be installed directly on any Linux based broadcast component on the iBiquity network by using the “farmer” utility. Log into the Linux machine as root or a super-user and from a command prompt type:

>> farmer -2 –m tx

and select the 4.3.0 baseline. The default platform type is an Exciter. To change the platform type before the application is started follow these steps from a Linux command prompt:

>> cd /usr/dab/irss-4.3.0/bin/tx

>> ./cmtx_set.sh –p1 –c8 (Exgines)

or

>> ./cmtx_set.sh –p 2 –c 1 (Exporter)

To start the application simply type “start” and select the 4.3.0 baseline. It is recommended that the “embedded” mode is used so that the application restarts automatically upon restarts or reboots. To configure the Linux machine into the set embedded mode type

>> setemb

The Linux machine will need to be rebooted for the changes to take place.

4.3.1 Configuration

Prior to using the system, each broadcast component must be configured properly so that accurate time synchronization can be achieved. To configure the Exgines select Platform->Link configuration from the main screen and a GUI similar to Figure 4-2 will be displayed. Select the Multicast enabled box, as shown in Figure 4-2, and hit OK. You will be prompted to reboot, choose OK; the system must be rebooted for the change to take effect.

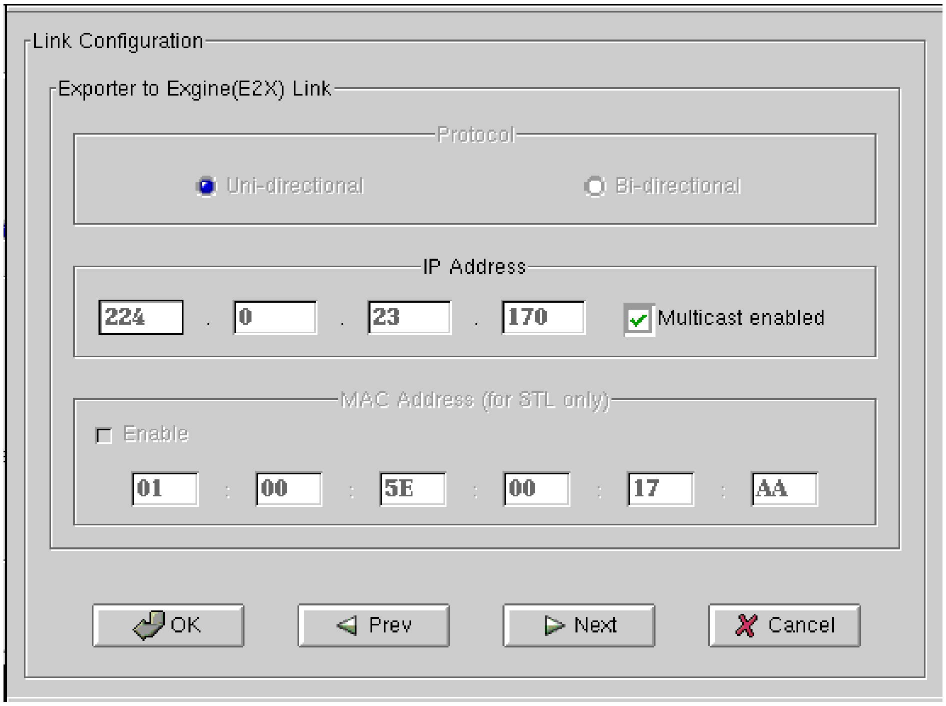

Figure 4-2 Exgine Link Configuration

To Configure the Exporter first select the "Audio" tab from the main screen and then select the "Analog Audio Diversity" tab and a screen similar to Figure 4-3 will be displayed.

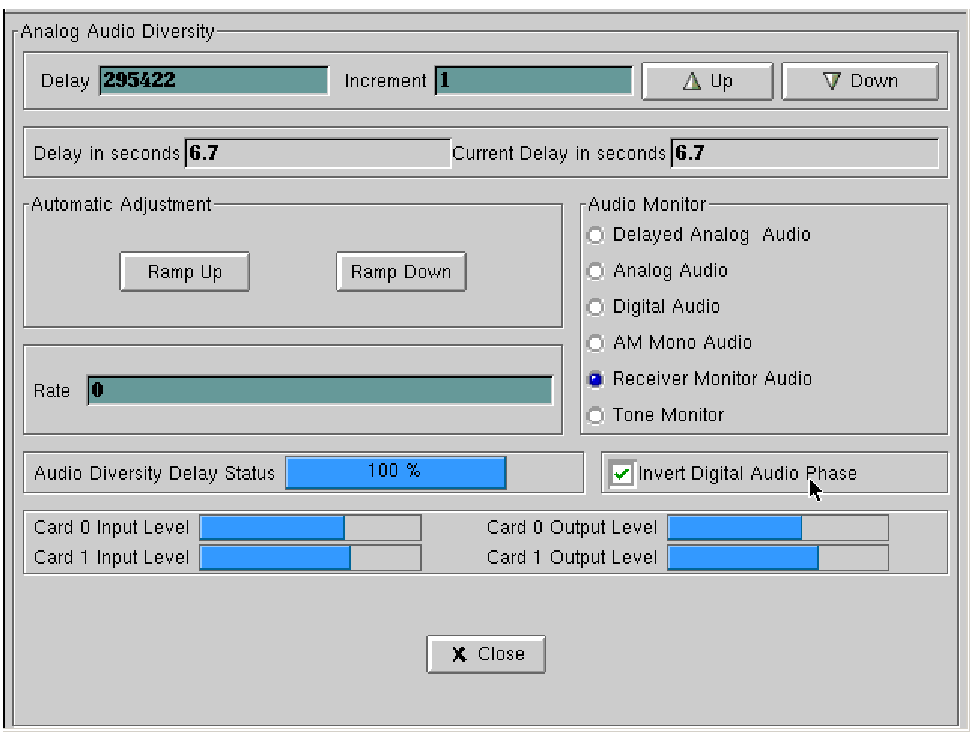

Figure 4-3 Exporter Analog Diversity

Check the box labeled "Invert Digital Audio Phase" and set the Automatic Adjustment Rate to zero as shown in Figure 4-3. Press "Close" to return to the main menu.

From the main screen select the "System" tab and then the "Waveform Synchronization" button and a screen similar to Figure 4-4 will be displayed. In the Synchronous Start group box select the "Enable" radio button (as shown) and then the close button. The system will prompt you for a system reset. At this time select "No" and you will be returned to the main screen.

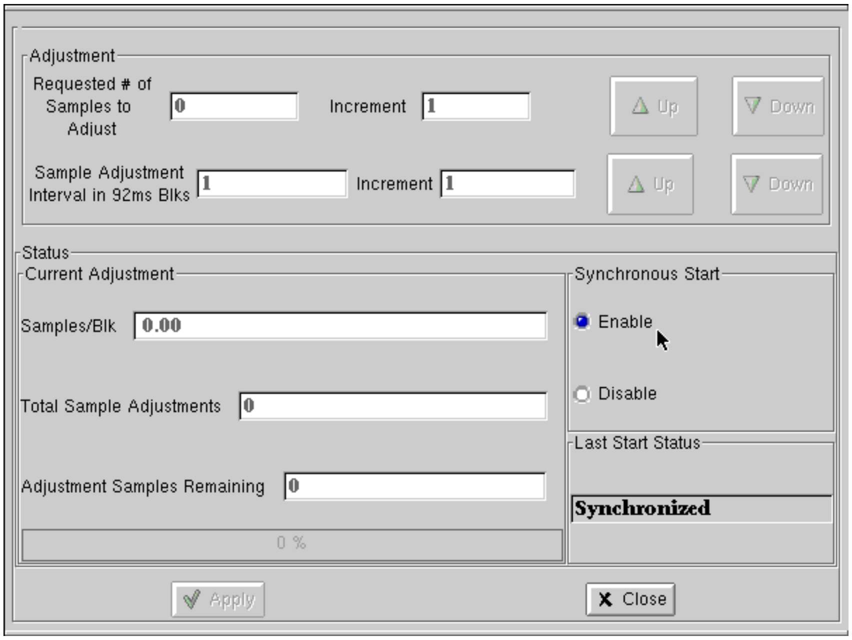

Figure 4-4 Waveform Synchronization

From the main screen select the "Platform" tab and then "Link Configuration" an a screen similar to Figure 4-5 will be displayed. Make sure you are on the "Exporter to Exgine (E2X) Link" and select the "Multicast enabled" check box. Then select "OK". At this time the system will ask for a System restart. Select Yes and the system will be restarted.

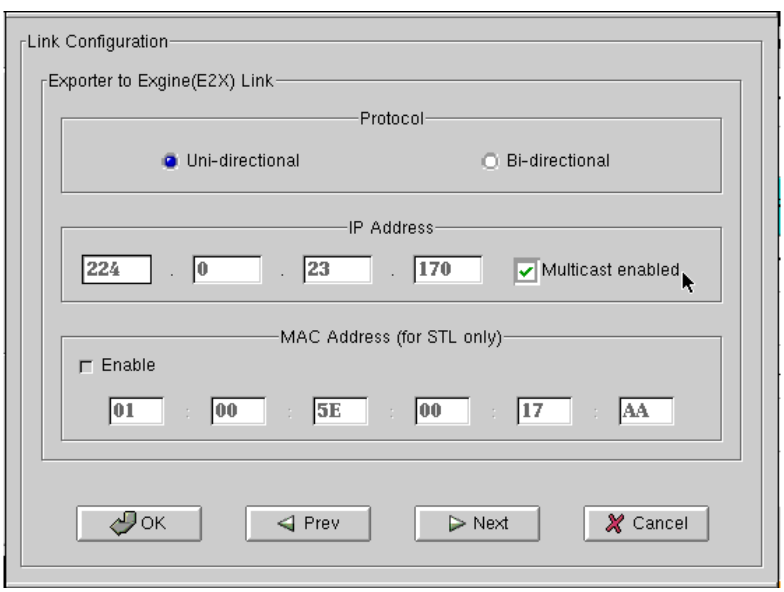

Figure 4-5 Exporter Link Configuration

To set the system in a multiple unicast TCP configuration requires careful hand manipulation of configuration files. If TCP connections are required see Reference [2] for a description of how to configure the system for TCP mode.

4.4 Measured Parameters

The digital oscilloscope is used to measure the time difference between the two Exgine output IF signals. The system will be started (synchronously and asynchronously) numerous times and the delay between the 2 signals will be recorded. This process will be repeated for a number of different network conditions. The digital oscilloscope will also be used to measure the “exactness” of the two waveforms by taking the difference between the 2 input channels and adjusting the scope time delay until the difference is minimized. However, since the oscilloscope only uses an 8-bit A/D converter, the accuracy of the measurement will be limited to ~40 dB.

For receiver performance, only audio block error rates will be recorded for varying time delays and signal level ratios. Although there are many other receiver performance metrics that can be recorded, the true measure of the system is weather or not digital audio is available and stable. If need be, future tests can be performed to obtain a more accurate understanding of receiver performance.

5 Test Plan and Results

5.1 Overview and Summary

Efficient SFN operations rely on three fundamental system requirements: 1) a high tolerance in corresponding sub-carrier frequencies and phases, 2) synchronization of signals in a time domain, and 3) exact symbol matching for corresponding sub-carriers. iBiquity's preliminary tolerances for SFN/Booster support are 20 Hz in frequency, 75 msec in time synchronization with a target of 10 msec and "bit-exact" symbol matching.

This section presents the lab tests and procedures that verify the HD Radio broadcast system meets or exceeds the tolerances specified above. Section 5.7 demonstrates the frequency accuracy is less than 0.5 Hz while Sections 5.2 and 5.3 demonstrate the time synchronization is less than 0.160 msec with a symbol matching that is greater then the limit of the equipment measurement capabilities.

5.2 Synchronous Start Alignment

The purpose of these tests is to make sure the whole system can be started and aligned repeatedly and to measure the maximum misalignment.

This test is performed by shutting down and restarting each component (Exporter and Exgine) multiple times and measuring the alignment offset. The Exporter software contains an option to also restart the Exgine software as part of the restart process, guarantying the Exgines are restart first and are in a ready state before the Exporter starts sending data. The process was repeated 20 times for both UDP multicast and TCP multiple unicast connections.

5.2.1 Results

The results for the synchronous start tests are shown in Table 5-1. As can be seen the measured misalignment is much less than the required 75 µsec and even less than the desired 10 µsec.

The peak to peak waveform differences for these tests were approximately 500 – 700 µV. The output of the Exgines are -20 dBm which translates into 0.022 volts RMS or 0.06 volts peak to peak so a 500 - 700 µV difference signal represents approximately a -41 dB to -38 dB which represents the limit of the equipment measurement capabilities.

Table 5-1Synchronous Start Results

	UDP (multicast)	TCP (multiple unicast)
Number of Trials	20	20
Average Misalignment (nsec)	-1.128	-17.2
Standard Deviation (nsec)	43.1	56.5
Maximum Misalignment (nsec)	79.8	93.7

5.3 Asynchronous Start Alignment

The purpose of these tests is to make sure each node in the SFN can be stopped and started independently of the other nodes.

These tests are performed by restarting a single Exgine of the SFN multiple times and measuring the alignment offset. The process was repeated 20 times for both UDP and TCP connections.

5.3.1 Results

	UDP (multicast)	TCP (multiple unicast)
Number of Trials	20	20
Average Misalignment (nsec)	7.42	10.8
Standard Deviation (nsec)	35.6	51.7
Maximum Misalignment (nsec)	79.2	160.8

5.4 Long Term Alignment Stability

Since each node must use a separate but locked time base, the purpose of theses tests is to make sure the system stays aligned over long periods of time, where long periods of time are on the order of days. This will be more a test of sustained GPS locking issues and potential limitations of the separation of time bases. Therefore, the digital oscilloscope will also be used to verify the 1-pps signals out of the GPS unit remain aligned.

5.4.1 Results

The system was run in both TCP and UDP modes for at least 2 days and no evidence of misalignment was observed even though the oscilloscope demonstrated some drift between the 10 MHz and 1 PPS signals in the two Exgines.

5.5 Adjustable Delay Performance

The purpose of these tests is to verify the accuracy and functionality of the Exgine slip buffer. Three different tests were performed.

1) The same sample slips were applied to both Exgines and the waveform are alignment was verified. Then the sample slips were removed from the Exgines and the waveforms were verified to realign. This was done for at least ten different values both positive and negative in the range of the slip buffer implementation (±17280 samples).
2) A small number of sample slips, both positive and negative, were applied to a single Exgine. Using an oscilloscope the slip in time between the 2 Exgine waveforms was verified to be in units of approximately ±1.344 µsec. Then the sample slips were removed and it was verified the 2 Exgine waveforms realigned.
3) To check the accuracy of large sample slips, the fact the there is a rational ratio between the FM sample rate and the audio sample rate was used to determine the delay introduced by the slip buffer. The MAM was used to make sure the analog and digital signals of a single Exgine are aligned to within 1 audio sample. Then the sample slip buffer was adjusted in units of 135 samples and the MAM should measure the analog/digital alignment to approximately units of 8 audio samples for each 135 sample slips applied to the buffer.

5.5.1 Results

For test 1) the delays measured were identical for both Exgines and they returned to known positions after the delays were removed. For test 2), Table 5-2 shows the measured delays using the oscilloscope. Only ±3 samples could be tested given the range of display of the oscilloscope. The errors are consistent with the 40-1 up sampling being performed within the DUC.

Table 5-2 Measured delays of single sample slips

Samples Delayed	Expected delay (μsec)	Measured Delay(μsec)	Error(μsec)
+1	1.344	1.310	0.034
+2	2.687	2.710	-0.023
+3	4.031	4.020	0.011
-1	-1.344	-1.310	0.034
-2	-2.687	-2.710	-0.023
-3	-4.031	-4.020	0.011

For test 3), Table 5-3 shows the measured audio misalignment as a function of sample slips. The MAM is not expected to be extremely accurate because it is using real audio content to perform the correlation, but it can determine if there is any growth in the error as the numbers of sample slips are increased or decreased. As can be seen from Table 5-3 this is not the case.

Table 5-3 Measured Sample Slip delays in terms of audio misalignments

Samples Delayed	Expected Audio Sample Misalignment	Measured Audio Sample Misalignment	Error*
0	0	0.46	-0.46
135	-8	-7.23	0.31
270	-16	-14.6	0.94
405	-24	-24.0	-0.46
1755	-104	-103.5	0.04
12150	-720	-719.9	-0.36
-135	8	8.3	0.16
-270	16	17	-0.54
-405	24	24.4	0.06
-1755	104	105.6	-1.14
-12150	720	721.5	-1.04

*The errors have been corrected for the initial misalignment.

5.6 Network Effects

The purpose of these tests is to make sure the system alignment is robust to various network impairments and to determine the tolerance to these impairments; making sure they match the expected performance from the software design. There are many different types of network impairments that can be applied to the various links between broadcast system components. Most of these are not a function of SFN performance, but of fundamental system performance and will not be addressed here. The only network impairments that will be addressed are network delay and throughput.

5.6.1 Delay

According to the software design, the system should be able to tolerate approximately a 500 msec delay between nodes. To verify performance, various delay settings were configured to determine the system failure point. The failure point was determined by performing synchronous start-ups and measuring the signal offset error to make sure it is consistent with the zero network delay scenarios. The delay is then systematically increased until the system is no longer able to perform a synchronized start-up. Once the limit of functionality is determined the delay is backed off slightly and the system is run over a long period of time to test the system stability to that delay. Ten trials at each delay were performed and the results are shown in Table 5-4. As can be seen from Table 5-4 the system is tolerant of network delays of up to 500 msec between nodes.

In the 600 msec case the system is still operational the two nodes just won't be synchronized. The software is designed to attempt a synchronized start for 2 minutes, if appropriate start geometry is not found for whatever reason (not just because of delay), the system will start in a non-synchronized mode, as indicated on the Waveform Synchronization GUI at the Exgine (see Figure 4-4).

Table 5-4 Measured misalignment as a function of network delay in multicast mode

Delay (msec)	Average Misalignment (nsec)
0	7.4 ± 35.6
200	11.7 ± 70.9
300	18.1 ± 69.2
400	29.1 ± 78.0
500	-7.6 ± 48.0
600	Not able to synchronize

There is interdependency on how much delay can be tolerated verses how much jitter can be tolerated. There exists a software configuration parameter in the system that can be used to optimize the trade-off between tolerances of network delay verse network jitter. The software parameter is used to adjust when the audio cards are started with respect to an ALFN. The closer the audio cards are started to an ALFN the more delay through the network can be tolerated but the less jitter can be tolerated because the size of the DUC buffer is reduced. We believe the parameter is set to an optimum value but if network conditions are encountered with severe jitter or massive amounts of delay, this parameter can be adjusted to potentially overcome these adverse network conditions. See Reference [2] for a complete description of this software configuration parameter.

In TCP multiple unicast mode, the TCP connection time out parameter is set to 300 msec. So any one way delays above 150 msec will cause a failure in the initial TCP connection. The value of 150 msec was determined so that a retransmission of a lost packet could occur and still not empty the DUC buffers.

5.6.2 Throughput

In the MP1 mode the E2X link requires an average throughput of 124.1 kbps. However, because of the existence of the large P1 packet (19441 bytes), and the 500 msec time-out that exists in the E2X network software, the minimum bit-rate that can be tolerated is approximately 300 kbps.

In order to verify the expected system performance the bandwidth of one of the nodes was systematically increased from 100 kbps to a point where the system is able to perform normally. For each different bandwidth value the system was synchronously restarted and if it is operational, the system was run for at least one hour to verify stability. In addition, the system was run over night at the lowest operationally valid bandwidth to insure long term stability.

The results of these tests are shown in Table 5-5 and Table 5-6, and as can be seen the system performed as expected.

Table 5-5 Multicast throughput test results

Bandwidth (kbps)	Results
100	The Exgine with the reduced bandwidth failed to become operational
150	Successful Synchronized start is achieved, and the system appears operational, but the E2X link is constantly being re-established.
200	Successful Synchronized start is achieved, and the system appears operational, but the E2X link is constantly being re-established.
250	Successful Synchronized start is achieved, and the system appears operational, but the E2X link is constantly being re-established.
300	Successful synchronized start and the system runs for extended periods of time*
350	Successful synchronized start and the system runs for extended periods of time*

*Dropped packets occurred about once every 4-8 hours, but after investigation it was determined the network emulation hardware was dropping the packets.

Table 5-6 TCP multiple unicast throughput test results

Bandwidth (kbps)	Results
100	The Exgine with the reduced bandwidth failed to become operational
150	The system starts and appears operational but many block count mismatches occur because of the delayed packets
200	The system starts and appears operational but many block count mismatches occur because of the delayed packets
250	The system starts and appears operational but many block count mismatches occur because of the delayed packets
300	Successful synchronized start, but the system does not run for extended periods of time due to the extra overhead of the TCP protocol
350	Success synchronized start and the system ran for over 2 days without an issue

5.7 Frequency Accuracy

The SFN system requirements calls for the HD Radio broadcast system to be a Synchronization Level I Transmission Facility. Since the iBiquity reference system was used as the basis for the requirements it should by default meet these specifications if the various EASU clocks are GPS locked. However, just to be sure, the system verification test “FM Broadcast Time and Frequency Accuracy” test (see Reference [4]) was performed to verify the accuracy of the carrier frequency.

5.7.1 Results

The results from the test are shown in Table 5-7 and in Figure 5-1 and Figure 5-2.

Table 5-7 Frequency Accuracy Test Results

Frequency Tested	Measurement	Tolerance	Pass/Fail
97.9 MHz	97.9000004	0.5 Hz	Pass
10 MHz	10.000000000	0.05 Hz	Pass

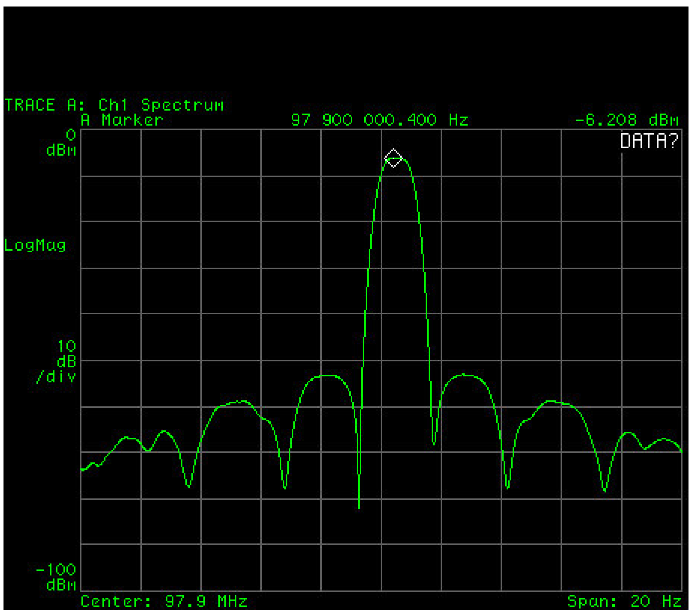

Figure 5-1 Center Frequency Accuracy Test

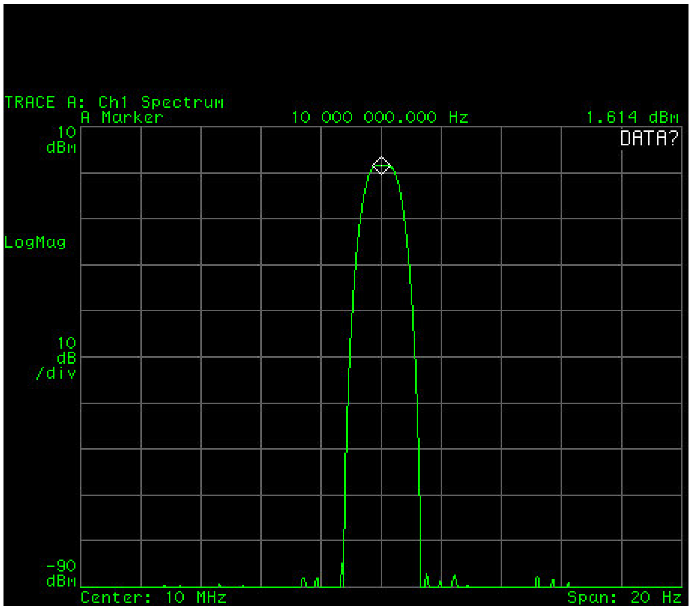

Figure 5-2 10 MHz Reference Clock Accuracy Test

5.8 Receiver Performance

The purpose of these tests is to get a sense of a receiver's performance as a function of the relative delay between the 2 nodes and to some extent signal strength. Since the only tests performed are in a flat, or non-fading, signal environment, they are not very useful for predicting actual performance. They are merely intended to provide an initial indication that the system is behaving properly so that the next level of testing (either more extensive lab tests or field tests) can commence.

For these tests the Exgine slip buffer was used to alter the delay between the two nodes of the SFN and the Audio block error rate was measured on both a 352-based IDM and a Linux receiver. The delay was systematically increased until the receivers were no longer able to maintain signal lock. At this point the signal strength of one of the nodes is reduced until the audio block error rate is again 0 for at least 1 hour.

It is expected that the system should work normally until the delay is ~ 75 µsec and be unaffected by the delay at a D/U ratio of ~4 dB [3].

5.8.1 Results

Table 5-8 and Table 5-9 show the results of these experiments. As can be seen both receivers perform as expected.

Table 5-8 Audio BLER as a function of delay (0 dB D/U)

D/U = 0 dB Delay (µsec)	BLER	
	Linux	IDM
< 65.8	0.0	0.0
67.2	8x10-5	2x10-4
73.9	2x10-5	8x10-5
75.3	5x10-4	2x10-3

76.6	5x10-4	2x10-4
77.9	No Lock	No Lock

Table 5-9 Audio BLER as a function signal strength (78 µsec delay)

Delay = 77.9 µsec	BLER	
D/U (dB)	Linux	IDM
0	No Lock	No Lock
1	No Lock	No Lock
2	No Lock	No Lock
3	No Lock	No Lock
4	0.0	0.0