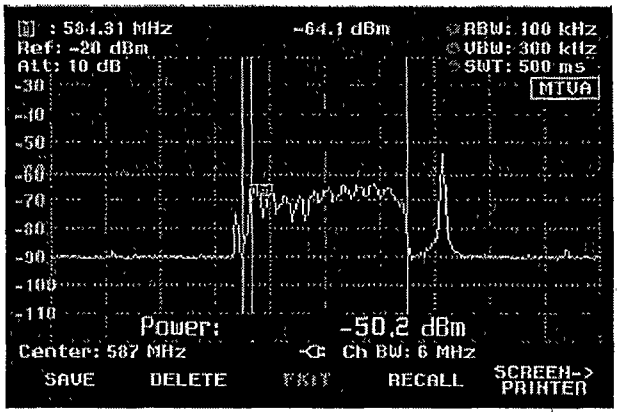

Figure 14a Example 7 spectrum analyzer plot

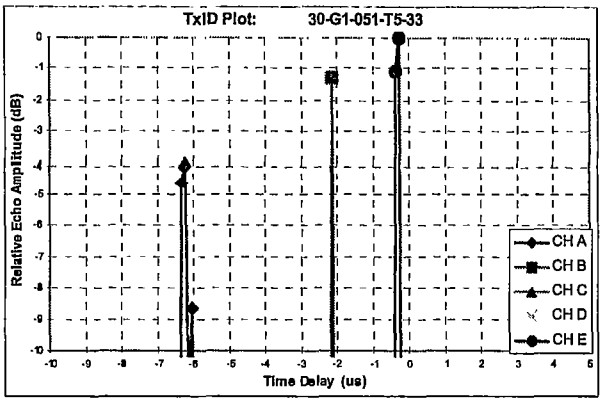

Figure 14b Example 7 TxID RF watermark analyzer plot

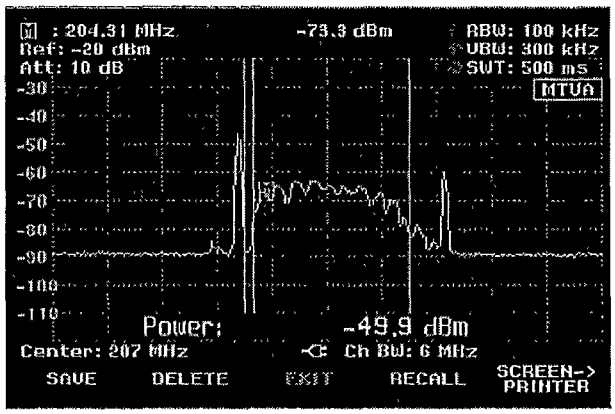

Figure 15a Example 8 spectrum analyzer plot

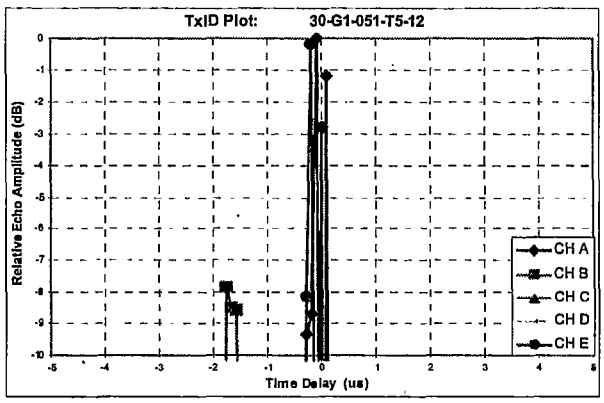

Figure 15b Example 8 TxID RF watermark analyzer plot

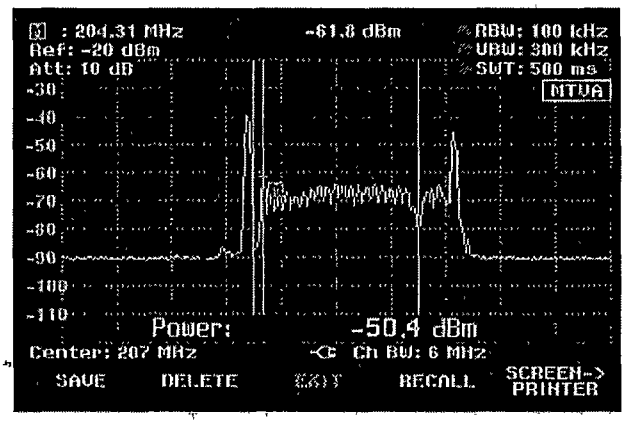

Figure 16a Example 9 spectrum analyzer plot

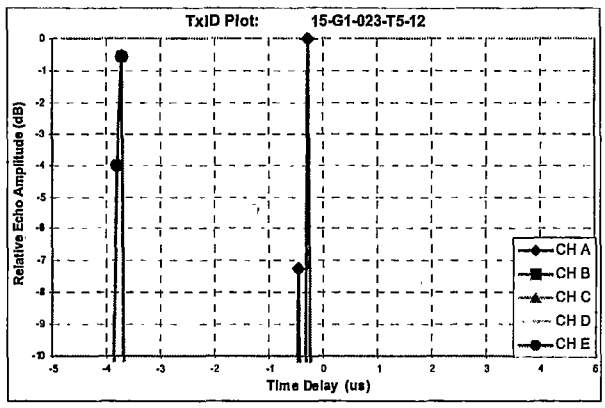

Figure 16b Example 9 TxID RF watermark analyzer plot

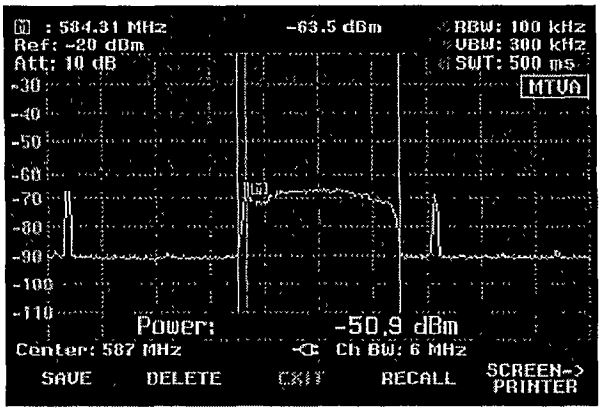

Figure 17a Example 10 spectrum analyzer plot

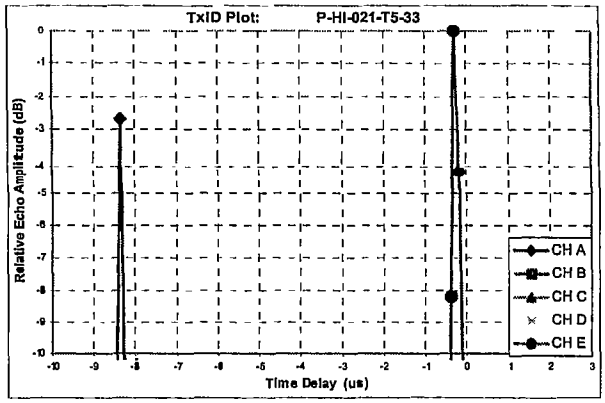

Figure 17b Example 10 TxID RF watermark analyzer plot

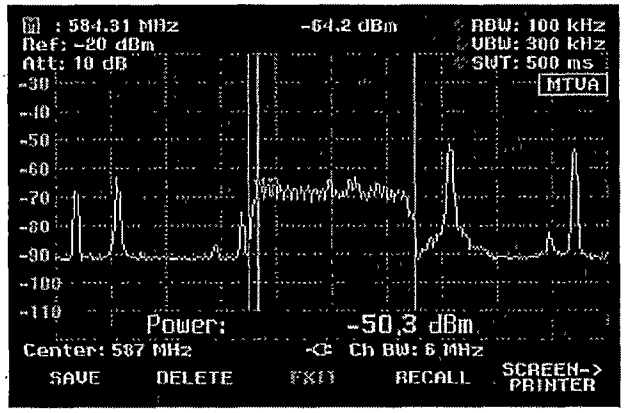

Figure 18a Example 11 spectrum analyzer plot

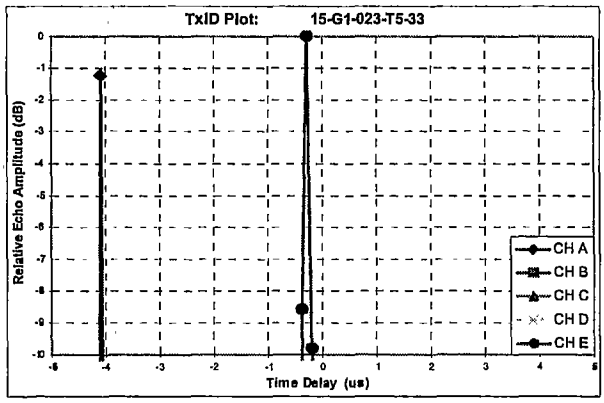

Figure 18b Example 11 TxID RF watermark analyzer plot

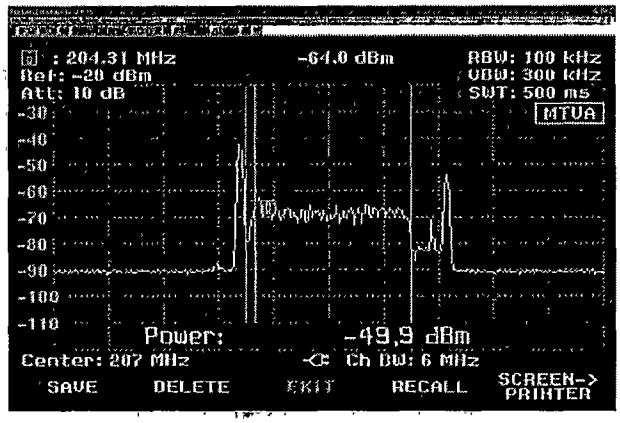

Figure 19a Example 12 spectrum analyzer plot

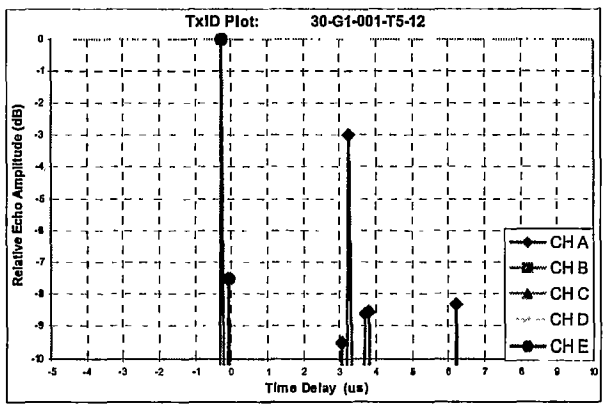

Figure 19b Example 12 TxID RF watermark analyzer plot

APPENDIX 1 DISTRIBUTED TRANSMISSION THEORY

Abbreviations and Definitions

CWDM **Coarse Wavelength Division Multiplexing** is a term describing the use of different wavelengths of light to transmit different signals in parallel on a single fiber. Coarse WDM uses widely spaced wavelengths as compared to the spacing of wavelengths in Dense WDM (DWDM).

DOCR **Digital On-Channel Repeater** is a term describing a system that receives digital signals over the air and retransmits them on the same channel on which they were received.

DTS[1] **Distributed Transmission System** is a term that was adopted by the Federal Communications Commission to describe the type of transmission method that it was proposing to license on a routine basis in a Notice of Proposed Rulemaking (NPRM).

DTx **Distributed Transmission** is a term describing a *form* of SFN that uses multiple transmitters fed over a Studio-to-Transmitter Link (STL). These transmitters separately generate the signals that they transmit in synchronization with one another and are controlled by data that they receive in common from a DTxA. The term DTx and all of its extensions were developed for and used in the ATSC standard (A/110B) and recommended practice (A/111) that describes operation of multiple-transmitter networks.

DTxA **Distributed Transmission Adapter** is a term describing a device that provides synchronization and control of multiple transmitters in a network by developing and inserting into the MPEG-2 transport stream feeding those transmitters additional data to which the transmitters respond by operating synchronously with one another and by emitting their signals with controlled time relationships.

DTxN **Distributed Transmission Network** is a term describing a form of SFN in which the transmitters are fed with signals from a DTxA over Studio-to-Transmitter Links (STLs), generate the radio frequency signals to be transmitted locally and emit them in synchronization and in time relationships with the signals from other transmitters in the SFN, under control of the data in the DTxPs from the DTxA.

DTxP **Distributed Transmission Packet** is a term describing an MPEG-2 transport stream packet that is used to carry synchronization, timing, and control information from a DTxA to one or more DTxTs.

DTxR **Distributed Translator** is a term describing a type of digital transmitter that receives signals *over the air*, converts them to a different channel, and retransmits them on that channel in synchronization with other DTxRs that share the output channel, all of which DTxRs respond to information in DTxPs directed to them.

DTxT **Distributed Transmitter** is a term describing a digital transmitter that receives data over a Studio-to-Transmitter Link (STL) and modulates and emits that data in synchronization with and according to parameters and instructions that it receives from a DTxA via a DTxP.

E-DOCR **Equalizing Digital On-Channel Repeater** is a term describing a form of DOCR that includes an adaptive equalizer in its receiver that has the purpose of minimizing effects of multipath in the channel from the source transmitter to the E-DOCR and of decoupling the receiving antenna from the E-DOCR transmitting antenna, thereby reducing the isolation required between those two antennas for a given radiated power.

Gap Filler A **Gap Filler** is a transmitter that is used to fill in the service area of a larger transmitter that is obstructed by terrain, man-made objects, or deficiencies in its performance from an area that it could otherwise serve.

Repeater A **Repeater** is a transmitter that receives signals from another transmitter and retransmits them. If it retransmits the signals on a different channel than the one on which they were received, it is termed a **Translator**. If it retransmits the signals on the same channel as the one on which they were received, it is termed an **On-Channel Repeater** (or, in the analog domain, a booster.)

SFN Single-Frequency Network is a term describing a network of transmitters sharing a single channel and jointly providing service to a common area. In order to minimize interference with one another, the SFN

[1] It should be noted that in earlier documentation prepared by and for the MTVA, the term DTS was used somewhat ubiquitously to describe many of the techniques related to Distributed Transmission, as did the FCC in its NPRM on the subject. This document limits the use of the term DTS to those places where the words Distributed Transmission System are meant and uses other abbreviations for other concepts in order to more *precisely* express what is meant. This modified usage in no way changes the intent of the earlier documents to describe techniques involving multiple synchronized transmitters.

transmitters operate in synchronization and with specific time relationships between their output signals. Both DOCRs and DTxTs can be used to form an SFN, and when DTxRs are used, they form an SFN on their common output channel.

General Terrestrial DTV Background

The Advanced Television System Committee (**ATSC**) digital television (**DTV**) standard (**Ref A1-1**) has been in existence since November 28, 1995 when the FCC's Advisory Committee on Advanced Television Systems (**ACATS**) *recommended* it as the U.S. terrestrial television standard. The specific terrestrial *transmission* system selected for the *digital* ATSC system was eight-level vestigial sideband modulation (**8-VSB**), which carries a 19.393 Mbit/second data stream. The ATSC system was ultimately codified into rules (**Ref A1-2**) by the FCC on December 24, 1996. At that time, the primary focus of the new DTV standard was implementation by the traditional method of using a single high-power transmitter, which was employed for many decades prior to that for both the analog National Television Systems Committee (**NTSC**) black and white and color television systems.

However, these new terrestrial DTV signals, while carrying digital information, are inherently analog RF signals. This is due to their analog sinusoidal RF carriers and their random, noise-like data modulation envelope that is spectrally-shaped by a steep root-raised-cosine band-pass filter (i.e., a large amount of signal ringing occurs). This means that these DTV RF signals suffer from all the same propagation effects that analog RF signals do.

DTV signals also suffer from all the same high-power transmitter and low-level receiver hardware impairments such as linear and non-linear distortion as well as reflections during propagation. Likewise, they also experience RF interference effects from analog NTSC signals and other digital ATSC signals. To the extent that it is possible, it is always best to combat these effects in terms of the broadcast transmitter design (e.g., transmitter location, antenna height, antenna azimuth and elevation patterns, beam tilt, polarization selection, radiated power, etc.).

All DTV receivers have certain RF performance characteristics. They all have front-end tuners that provide proper tuning of desired channels (CH 2 – CH 69) and convert the incoming signal to an intermediate frequency (**IF**). These tuners also determine both the sensitivity (low-level) characteristics and the overload (high-level) performance characteristics of the DTV receiver.

Likewise, there is a *linear* adaptive equalizer present in *each* DTV receiver whose function is to remove any and all *linear* distortion from the signal, such as hardware-induced frequency magnitude and group delay distortion or propagation-induced multipath distortion. However, the amount of linear distortion (e.g., amplitude and group delay) that can be removed by the equalizer is always limited. Most modern day DTV equalizers can remove essentially 100% echo amplitudes if the delay is not too great (e.g., close-in echoes that are less than a few microseconds), and can now handle *single* echo delays (e.g., delay spread) from at *least* -25 μsecs to at *least* +45 μsecs. Many of the most recent DTV equalizers have variable-position center taps that can be intelligently moved by smart algorithms depending on the presence of pre-echoes and post-echoes. Also, they can handle fast *dynamic* multipath caused by Doppler effects from moving reflective objects. The importance of the cancellation delay range of both pre-cursor and post-cursor multipath can not be stressed enough, not only for severe multipath distortion experienced in major urban areas where typically no line-of-sight exists to the single transmitter, but also for Single Frequency Network (**SFN**) and Distributed Transmission (**DTx**) designs, as will be discussed below.

The most overarching characteristic of DTV reception is the digital "cliff effect" which is caused not only by the nature of the digital transmission system itself, but also by the extreme amounts of forward error correction (trellis-coded modulation and Reed-Solomon coding) that are included in the ATSC system. White Gaussian noise is present at the front end of every tuner, with the amount of noise (i.e., noise floor) dependent on the tuner noise figure (**NF**). As the incoming DTV signal decreases to a level that causes a signal-to-noise ratio (**SNR**) near 15 dB, the video and audio outputs of the DTV receiver change from error-free perfect pictures and sound to all-error frozen pictures without sound in a matter of less than 1 dB of signal change. This is a drastic occurrence, which is why it is called the digital "cliff effect" in DTV broadcasting.

After the digital television standard was selected in 1996, the FCC created a table of channel allotments in 1997 for the U.S., loaning every eligible *full-service* television station (about 1600 in the mid 1990s) a second 6 MHz RF channel with which to transmit DTV during the transition. However, not all stations were able to replicate their VHF NTSC coverage with a single high-power transmitter using their allocated UHF DTV channels, and some had local ordinances preventing them from putting up a tall transmitting tower that could have extended the optical and radio horizon and increased the number of line-of-sight receiving locations. Also, there was a growing expectation of improved outdoor *and* indoor television delivery systems for the 21st century as well as handheld and mobile systems. Therefore, other means of terrestrial television *distribution* were considered in order to provide improved over-the-air (**OTA**) television service.

Before describing various digital service extension strategies, a brief review of the traditional "single stick" transmission methodology is in order. The required large effective radiated power (**ERP**) used by single transmitter systems often has more than 30 dB of margin for line-of-sight conditions. However, *non*-line-of-sight conditions that occur at the edge of

service (e.g., radio horizon), in rural areas with hilly terrain, or in urban areas with tall buildings require this extra 30 dB of transmitted power to overcome the greater attenuation and signal level variability (i.e., fading) in these propagation paths. With the digital cliff effect, a blank DTV picture 50% or 60% of the time is unacceptable. This is the reason that the FCC did *not* use its F(50, 50) statistical field strength prediction curves for DTV allocations which assume only 50% of the locations have acceptable field strength 50% of the time. They chose to use F(50, 90) prediction curves to get the 90% time variability that was desired for these new digital signals which exhibit cliff effect characteristics.

Additionally, RF propagation effects are even worse at UHF frequencies, which contain the largest number of DTV channel allocations, since UHF signals are "propagationally-challenged" in non-flat terrain due to their shorter wavelengths and their propensity to not bend as well as VHF signals. And when UHF signals do bend (due to refraction or diffraction), they can exhibit significant time variations that are dependent upon many different propagation path variables (e.g., temperature, weather, diurnal, seasonal, etc.). This condition, also called fading, requires a larger RF signal level in order to stay above a predetermined minimum field strength level that provides acceptable DTV service (i.e., for an acceptable percentage of time).

As stated above, this desired increased reliability for DTV service is important at both the noise-limited contour far from the transmitter and in highly urbanized areas that have very large structures close to the transmitter (e.g., urban clutter). In these cases, the signal path is either blocked or there are significant reflected signals present. To minimize these effects, transmitter height above average terrain (**HAAT**) is much more important to broadcasters than effective radiated power since increased height provides more line-of-sight (or near-line-of-sight) receiving locations while increased power can potentially cause interference into neighboring signals.

Therefore, a need exists to provide DTV reception in areas affected by the signal impairments discussed above. Some form of television service extension is often necessary to "fill in the gaps" where poor *coverage* and/or poor *service* exists. The primary purpose of these so-called "gap fillers" is to provide increased signal *levels* or improved signal *quality* to areas that are deficient. Care must always be exercised, however, to avoid increased co-channel or adjacent channel interference to neighboring analog or digital channels in the gap filler service areas.

Service Extension Techniques

Historically, a number of techniques have been applied by broadcasters and others to address problems with reception of *analog* NTSC broadcast signals in areas beyond the reliable service areas of single, high-power transmitters such as in areas where these signals are blocked by terrain or man-made objects. Analog service extension to these "propagationally-challenged" areas has involved both "translators" and "boosters."

Translators receive, amplify, and re-transmit the analog signals from a main transmitter (or from other cascaded translators earlier in a chain) on a different RF channel from the one received, as shown in **Figure A1-1**. **Boosters**, on the other hand, receive, amplify, and retransmit signals from a main transmitter (or from a translator) on the *same* RF channel from the one received, as illustrated in **Figure A1-2**. The FCC Rules provide for these two types of service extenders for use with NTSC signals. As a practical matter, boosters have been very difficult to implement, even on UHF channels, because of the difficulties of keeping transmitted signals from feeding back into receiver inputs. This unwanted feedback causes severe signal distortion, which limits the power that can be radiated, and can lead to oscillation of the booster systems when inadequate isolation is provided between transmitting and receiving antennas. Consequently, translators have been far more widely used to repeat analog NTSC signals than boosters. However, their application has been *somewhat* limited by the lack of availability of additional channels in which to operate them, primarily due to the NTSC taboo allocation requirements.

With the advent of digital television transmission, some new techniques have become available for the extension of service areas and the overcoming of propagation challenges. Since digital transmission actually involves radiating analog signals that carry digital information that can be recovered by DTV receivers, the same techniques that are used for analog service extensions remain available for use with DTV. ATSC digital receivers universally contain adaptive equalizers that overcome the effects of multipath caused by the natural environment that would normally prevent successful reception. Advantage can be taken of the adaptive equalizer technique to permit designing "**single-frequency networks**" that re-use the spectrum on the same channel. This can be accomplished either through "on-channel repeaters" (the digital equivalent of analog boosters) or through use of multiple, *synchronized* transmitters sharing the RF channel and producing signals that impact receivers in the *same* way as multipath. However, the simplest and most cost-effective form of service extension is the *unsynchronized* translator, which is also currently the most commonly implemented type of service extension. In the new digital environment, finding channels is easier than in the analog-only environment since the new DTV signals can safely use the analog taboo channels if certain conditions are met (**Ref A1-3**). However, in many situations, available RF channels are still not available to be allocated to interested broadcasters. Therefore, in these cases, the preferred method of service extension is an SFN, which will be the focus of the remaining background material.

General Single Frequency Network Background

SFNs can be constructed with two fundamental types of transmitters: off-air repeaters and separately-fed repeaters. Off-air repeaters receive RF signals, provide minimal processing, and then retransmit them, either on a different channel or on the same channel as the received signal. Separately-fed repeaters require some sort of studio-to-transmitter link (**STL**) to each transmitter, such as microwave, fiber, or satellite.

Two specific types of off-air digital repeaters are possible for use in SFNs. The first type is a translator repeater that heterodynes (either directly or via an intermediate frequency) the received off-air signal to a new RF channel for transmission to shielded areas. These are called Distributed Translators (**DTxRs**) when more than one shares an output channel and their outputs are synchronized with one another. The second type is a Digital On-Channel Repeater (**DOCR**), which amplifies and redirects the output signal on the same frequency used by the off-air input signal.

Digital *translators* (synchronized or unsynchronized) are by far the easiest to implement, assuming there are free channels available for use in the desired reception area, and they are the best and first choice of broadcasters. The use of a *different* channel frequency makes the repeater design and implementation relatively easy. The input and output antenna requirements are simple since there is no problem with self-interference from the translator's output signal feeding back to its input, which allows stable operation (i.e., no output-to-input feedback interference, distortion, or oscillation problems). Translator operation is not limited to terrain-shielded coverage areas since the output and input signals can "overlap" each other without causing co-channel interference. Implementation on VHF channels (input or output) is essentially as easy as on UHF channels (input or output). And the implementation cost is relatively low compared to more sophisticated approaches. The use of a separate channel frequency for the repeated DTV signal does not cause tuning problems for DTV receivers because PSIP data alleviates viewer channel selection confusion. To add to all of these benefits is the fact that *digital regeneration* (VSB demodulation to transport stream (**TS**) with multipath cancellation and forward error correction before VSB re-modulation) is relatively inexpensive and provides a pristine low-power DTV output signal for retransmission to shielded areas, assuming that the incoming DTV signal remains above the threshold of errors. **Figure A1-1** illustrates a simple translator system. Note that even first adjacent channel signals can be eliminated and not inadvertently re-transmitted by these digital regenerators. The only downside to translators is the requirement for frequency allocation planning, that is, finding an *available* RF television channel in the desired service area.

In many rural areas, extra channels are available for translator service, especially when co-located translator sites exist on mountain tops. Co-siting of translator sites allows relaxed taboo channel requirements (**Ref A1-3**). Even in some large urban areas, extra channels are available for use with low-power DTV signals and *directional* transmitting antennas since the same translator channel can be used at opposite ends of a large metropolitan area by having the low-power transmitters pointing away from each other through the use of highly directional transmit antennas. Important translator design aspects consist of finding an optimum translator site, determining the required input signal level and signal quality, specifying digital regeneration equipment, selecting an output emission mask filter (simple or stringent mask, per the FCC's low-power television (**LPTV**) and translator rules), and determining the minimum required radiated output power (to avoid causing interference to neighboring signals and yet provide the desired coverage and service). The use of translators (synchronized or unsynchronized) is the simplest and most cost-effective means to provide gap filling, and is the broadcaster's first choice. However, there are many areas that do not have extra spectrum available, thereby forcing broadcasters to consider complete on-channel techniques.

Digital *On Channel Repeaters* are an alternate means of repeating the signal on the *same* channel frequency when no extra spectrum is available for translators to use, such as in frequency-congested urban areas. This signal booster technique has been very carefully used in the past with analog NTSC signals. On-channel repeaters, like translators, take their input signals from off the air, but instead of translating them to a different RF channel, they repeat an amplified version of the signal on the same channel. Not only is this spectrum efficient, but it even may allow reduced main transmitter power since the main signal doesn't have to reach the specific, propagationally-challenged areas that the repeaters are servicing.

There are *five* basic configurations of on-channel repeaters, each comprising a similar overall system design of a receiving antenna, a signal processor, a power amplifier, and a transmitting antenna. The system design variation comes in the type of signal processing that is performed:

1) Simple on-channel LC band-pass filtering (*shortest* transit delay, fraction of a microsecond)
2) Frequency translation to and from an IF frequency (e.g., 44 MHz), with a simple IF LC band-pass filter (*short* transit delay of less than 1 μsec)
3) Frequency translation to and from an IF frequency, with a surface acoustic wave (**SAW**) filter (*longer* transit delay of 2 – 3 μsecs)

4) Frequency translation to and from an IF frequency, with SAW filtering and *equalization* of multipath effects on the receiver input, combined with level slicing of the symbols to restore them to accurate amplitudes, enabling transmitter pre-correction and use of an RF watermark (*longest practical* transit delay, on the order of 5 μsecs)

5) Complete VSB demodulation to the transport stream, complete with equalization and error correction before re-modulation back to a pristine VSB signal on the same RF channel (impractically *long* transit delay, in milliseconds)

It should be noted that a disadvantages of using DOCRs is that they can cause self-induced co-channel (or multipath if a synchronized digital regenerator is used) interference in coverage regions that are *overlapped* by both the main and repeated signals since the same frequency is used for both. In each of the five configurations described above, the time delay through the on-channel repeater increases with more sophisticated signal processing. The possibility of causing this multipath interference to consumer DTV receivers essentially limits the on-channel repeater design to be of an *analog* type or one with digital equalization and level slicing (i.e., *not* a full digital regenerator, as described in #5 above), except in the case of *significant* terrain-shielding between the main and repeated signals (i.e., minimal or no overlapped coverage regions where self-induced repeater multipath occurs). Minimal-delay repeaters are required since excessive processing delay, such as exist in full digital regenerators, would create echo delays in overlapping coverage regions that are longer than the cancellation ranges of practical consumer DTV equalizer hardware. The analog and equalizing digital repeaters typically have only a few microseconds or less of filter delay (the larger delay times are often from a SAW filter in the analog case and are from a combination of a SAW filter and the equalizer in the digital case), and therefore re-transmit the signal very quickly so that shorter self-induced echoes exist in overlapping coverage regions.

It is well known that *analog* repeaters always degrade the signals they carry (i.e., they never make them better) since there will be an *accumulation* of circuit and propagation noise and linear/nonlinear distortion along any cascaded system, which limits the quality of the repeated signal. The advantage of *digital regeneration*, which equalizes the signal and removes data errors (with forward error correction) before re-modulation and transmission, is *not* practical in off-air on-channel repeaters due to their unacceptably long processing delays (except in the special case where there is essentially complete terrain shielding between the main and repeated signals). The recently developed alternative listed in #4 above uses equalization of the received signals both to minimize the effects of multipath from the source transmitters to the repeaters and to counteract the effects of any feedback that occurs from the repeater transmitting antennas to their receiving antennas. The equalization is combined with level slicing to eliminate the noise from the earlier parts of a cascade, but noise effects large enough to cause slicing to the wrong levels will build errors into the transmitted signals, thereby limiting operation to areas with DTV input signals providing reasonably large (>25 dB) signal-to-noise (**SNR**) values at the receiver's tuner input. In all five fundamental designs, the delays through the repeaters are always *positive* values determined by the particular designs. That is, the output signals will always be transmitted at a *later* time with respect to the input signals. That is, there is no way in a practical (i.e., real world) implementations to achieve a "negative delay," as would be required to *reduce* the effective time delays between overlapping signals in target service areas.

Of course, the possibility of output-to-input feedback interference and distortion (or possibly oscillation) exists with these on-channel repeaters, which requires very careful antenna system design (selecting antenna patterns and determining physical mounting of the antennas). Antenna patterns may be required to have very narrow beamwidths or extreme front-to-back ratios, and transmitting-to-receiving antenna isolation levels of 80 dB to 100 dB. Antenna isolation in all three dimensions (x, y, and z) is often very necessary, along with additional shielding techniques. Thus, on-channel repeaters are only practical at relatively low power (a few hundred to a few thousand Watts of effective radiated power) and at the shorter-wavelength UHF frequencies (***not*** at VHF frequencies) where good antenna isolation is possible.

On-channel repeaters require careful network designs, even more so than for translators. Selection of the repeater *site* is important to take advantage of any terrain shielding (minimize overlapping coverage region for acceptable self-induced interference), antenna height (best controlled pattern possible), and radiated power (minimal output power for desired coverage and minimal interference). The critical network timing is dependent on the geometry of the locations of the main signal source location, the on-channel repeater site, and the receive site, with any timing calculations assuming the signals are traveling at the speed of light ($3x10^8$ m/sec or about one mile in 5.4 μsecs). The signal transit delay through the repeater hardware then must be added to this propagation delay value. Any additional delay in the repeater forces the delay spread in the wrong direction, thereby *not* helping the consumer DTV receiver equalizers to remove self-induced multipath. Also, the repeater output frequency must be identical (or at least *within* 0.5 Hz) with that of the input frequency to avoid any generation of self-induced *dynamic* multipath (e.g., if translation is to an IF frequency for filtering before heterodyning back again to the exact same RF channel). And the output emission mask filter is critical to once again avoid any interference to existing nearby adjacent channels. **Figure A1-2** illustrates a simple on-channel repeater system. Note that *adjacent* channel analog (or digital) signals at the on-channel repeater input are troublesome for an on-channel repeater since they cannot be fully removed with filtering without significant linear distortion of the desired DTV signal. Therefore, for all the reasons stated above, including power and timing limitations, on-channel DTV repeaters are often *not* desired for broadcasters.

However, the undesirability of off-air on-channel repeaters does *not* preclude all on-channel or SFN techniques. Distributed transmission, commonly referred to as DTx, can avoid some of the unacceptable characteristics of on-channel repeaters, although with increased complexity. By delivering the digital data that is to be transmitted by a means *other* than over-the-air broadcast and by synchronizing the output signals of all the digital transmitters, acceptable DTV reception can be possible with careful system design.

Distributed Transmission Overview

Distributed transmission (**DTx**) technology for digital television is a method of providing DTV signal coverage and service to an area using multiple *synchronized* low-power transmitters on the *same* channel frequency. This is in contrast to the *traditional* analog television method of using a single high-power transmitter on a tall tower to accomplish signal coverage and service. ATSC distributed transmission systems (**DTS**) apply single frequency network (**SFN**) technology to the ATSC system. The ATSC has developed a synchronization standard (**Ref A1-4**) and guidelines (**Ref A1-5**) for DTx transmission that provides means for transmitter synchronization in *carrier frequency* and *symbol emission* as well as proper timing adjustments to minimize *timing differences* between signals emanating from multiple transmitters. However, the DTx transmitter output signal format must be *identical* to that of the 8-VSB standard (**Ref A1-1**), and therefore be completely backwards compatible. Others have written descriptions regarding DTx systems and their design (**Ref A1-5, Ref A1-6**). The ATSC transmission system design has been shown to be very flexible, lending itself to both SFN and Enhanced VSB (**E-VSB**) technology.

DTx is a method where DTV signal coverage and service can be increased over that from a single transmitting antenna by strategically placing these multiple *low*-power transmitter sites *within* the original DTV service area, especially in areas where there is a lack of tall tower space (i.e., reduced vertical "real estate") for a single high-power transmitter, or in spectrally-congested areas where there may be a lack of available RF channels for DTV translators. DTx is different from on-channel repeaters in that each DTx transmitter receives its input signal from a studio-to-transmitter link (**STL**) signal that is different from the over-the-air signals delivered to viewers. This alternate *delivery* path that is separate from the terrestrial *transmission* path provides the necessary flexibility to easily adjust the timing and amplitudes for optimal DTS performance, which is limited primarily by the consumer DTV receiver performance.

In the traditional single transmitter system, the cost to the broadcaster of reaching the last mile of coverage with a single high-power transmitter is the most expensive. For instance, at 50 miles from a UHF transmit site and using a 1000' HAAT transmit antenna, it takes approximately 1 dB for every additional coverage mile, or 3 dB for every 3 miles of extended coverage (**Ref A1-6**). That means the last 3 miles requires double the transmitter power output (**TPO**), which requires a significant initial hardware cost as well as on-going operational (electricity) cost. Likewise, RF interference to nearby analog and digital television signals is always an issue to consider. When using the FCC F(50, 10) field strength prediction curves for *interference* signals compared to the F(50, 90) curves for the *desired* service signal, a UHF DTV transmitter can cause interference at 3 times the radius of its service contour. This explains the large distance separations of high-power transmitters (**Ref A1-6**). Furthermore, indoor DTV reception, expected to be used in urban and suburban areas within 20 to 30 miles of the transmitters, requires stronger signal levels outdoors by as much as 20 dB or more to overcome lower gain from indoor antennas and increased signal attenuation from antennas at lower antenna heights above ground level and building attenuation loss. Achieving strong signals inside the viewer's home is not difficult if the reception site is relatively close to the transmitter tower, but it requires large amounts of transmitted power when strong indoor signals are desired 30 miles away.

Many of the challenging *RF propagation problems* as well as *RF spectrum availability* issues can be addressed with SFN technology now that consumer DTV receiver implementations (tuners and equalizers) have improved significantly since they were originally introduced in late 1998. Since SFN technology creates absolutely synchronized multiple transmitted signals, the presence of multiple DTx signals will primarily appear as quasi-static multipath to DTV receivers. Sometimes the levels of multiple synchronized transmitter signals can be comparable to each other at some reception sites in overlapping coverage areas, which mean that severe self-induced multipath would be present at the DTV receiver's input. However, with proper SFN design, the modern DTV equalizers can often converge on a solution to cancel or significantly reduce the multipath effects and provide error-free reception. As the equalizers continue to improve, SFN self-interference requirements will relax.

A number of benefits exist when multiple synchronized low-power transmitters are employed to cover large areas. It is easier to maintain more uniform, higher field strength coverage since the area that each low-power transmitter covers is smaller than that of a single high-power transmitter. This reduced coverage distance also reduces signal level variability with time (i.e., fading), which requires less fade margin and therefore a lower radiated transmitter power and/or transmit antenna height. The lower radiated power has the *potential* for reduced interference, although it depends on where the neighboring transmitters are located and the relative level of their RF signals compared to the DTx signals near any of the DTx transmitter sites. Terrain limitations are easier to overcome with multiple transmitters since the low-power transmitters can be strategically placed to fill in the nearby shaded regions better than the one tall tower that resides far away. In many instances, the terrain

limitations that are caused by obstructions such as mountains or large hills are, in fact, what may *facilitate* DTx network amplitude and delay design by minimizing overlapping coverage regions.

Another advantage of a distributed transmission network is that it provides a form of transmitter *diversity*. That is, the signal to any given DTV receiver can come from multiple transmitters from multiple directions over different terrain. This form of diversity reduces the chance of weak signal coverage areas or large propagationally-induced spectrum nulls due to multipath that might occur if only one high-power transmitter were used. As a matter of fact, inclusion of one high-power transmitter on the same channel along with all the low-power transmitters is *not* necessarily a requirement. Finally, when indoor reception is desired (either in urban or suburban areas), multiple transmitters may fill the houses with stronger and less time-varying (fading) signal levels, thus improving DTV reception reliability with simple receive antennas **(Ref A1-6).**

DTS methodology is also useful when signal coverage from a main transmitter may not be adequate to properly provide good enough signal *quality* across a large area. Reduced signal quality may be in the form of severe signal fading or significant multipath levels. Poor signal quality can certainly be caused by rough terrain (i.e., terrain shielding) out in rural areas, but it also can be caused by large buildings in a highly urbanized city like New York City or Chicago. The matter may be complicated if the main transmit site is located at a facility where the transmitting antennas cannot be placed in an optimum position, such as *some* of the DTV antennas on New York City's Empire State Building (**ESB**). In this particular situation, some of the transmitting antennas are currently side-mounted near the bottom of the tower structure that resides on top of ESB, and therefore they do not provide consistent coverage as was determined from fly-around and terrestrial field strength measurements.

Distributed transmission techniques can also be applied to *translator* systems, and these are called *distributed translator networks*. Traditional translator systems typically use many different channels, thus sometimes filling up the spectrum in a given region. In this scenario, there is no large main transmitter but rather only synchronized low-power distributed translators (**DTxRs**) all operating on the same channel. With multiple cascaded levels (tiers) of translators, only one *additional* channel is required (i.e., a total of two channels), where the signal is conveyed to each tier *not* through the traditional STL but rather through an over-the-air RF channel from the previous translator tier. One additional channel is required so that none of the distributed translators has to receive and transmit on the same channel, which would be a very challenging design as described in the previous section. Nevertheless, this is still a frequency efficient method to transmit digital signals over a wide area with challenging terrain. An interesting note is that the ATSC distributed transmission model did not originally support distributed translators, but that capability was added towards the end of the ATSC standardization process due to interest for rural area service (**Ref A1-7**). Many of the same DTx system requirements apply to this technology as well.

It is important to remember that distributed transmission technology is ***NOT*** on-channel repeater (booster) technology. The input to each transmitter is ***NOT*** taken off-the-air from the *same* channel on which it is re-transmitted, as described above for analog booster systems. Rather, a *separate* link is used to convey the data to the transmitter sites, which removes the tough requirement of antenna isolation that is required in order to prevent oscillation in booster applications. Even with the low processing delay time (less than a few microseconds due to RF and IF filtering) at each repeater site, it can be enough to prevent proper DTV receiver synchronization, and consequently degrade system performance significantly in overlapping coverage regions between the main and repeated signals. Rather, DTx uses a separate digital distribution channel to feed each transmitter site with the 19.393 Mb/sec MPEG transport stream. These separate feeder channels can be conventional microwave or fiber **STLs**, a satellite link, or they could be another television channel (e.g., distributed translators). Remember that when a main transmitter and some lower power transmitters (perhaps gap fillers) are used together in a network, the main transmitter is also a Distributed Transmitter because it receives, and is slaved to, the same signals over the STL just like all the other transmitters.

Therefore, a number of *low*-power distributed transmission transmitters (**DTxTs**) can be located in "cells" throughout a service area that can aid a main signal in providing DTV service to more viewers. Selection of the gap filler sites should be such that minimization of the overlap reception areas from the main and gap filler signals is achieved and yet DTV signal strength and quality to the largest area possible is realized. In a DTx network (DTxN) design, a few cells can each cover relatively large areas ("large cell" scheme) or a lot of cells can each cover a relatively small area ("small cell" scheme). In either case, while these cells may appear to be similar to a cellular telephone system, they are different in that all cells have the *exact* same signal on the *exact* same channel frequency.

From the information just presented, it should be obvious that DTx systems hold promise in those areas where both coverage and service need to be *extended* but no channels are available for simple translators. However, great care must be employed to design the system so that DTV receivers have the best chance for successful reception. This means accounting for both the DTS hardware parameter optimization, and considering also the impact that DTx might have upon consumer DTV receiver performance. Therefore, an important issue to consider all of the ***limitations*** in distributed transmission systems.

For example, the self-induced interference that occurs in the overlapping coverage areas from two or more DTx signals comes in the form of multipath since all the DTxTs are synchronized to transmit the exact same *symbol sequence* on exactly

the same RF *carrier frequency*. Therefore, the ability of DTV receivers to handle potentially severe multipath is critical to the success of DTx systems, which means that certain DTV equalizer performance characteristics are an important consideration.

In a DTx environment, a DTV equalizer should have the ability to cancel one or more large amplitude (near 0 dB / 100%) *quasi*-static pre-echoes and post-echoes over a reasonable delay spread (e.g., at *least* ±5 μsecs). For smaller but still significant amplitude echoes, an even larger cancellation delay spread should be possible (e.g., at least ±20 μsecs if not ±30 μsecs). Equalizers should even be able to handle multipath conditions that *alternate* between large pre-echoes and post-echoes (called a "bobbing channel"), a very challenging condition. These conditions can commonly occur in rural areas behind hills, in urban "concrete canyons", or with indoor reception, and are the result of absolutely no line-of-sight between the receive and transmit antennas. Therefore, in DTV receivers, both the echo delay time cancellation window and the echo amplitude cancellation window are important, with a typical performance scenario of reduced echo amplitudes that can be cancelled with increasing delay. Such an echo-amplitude versus echo-delay profile curve exists in the ATSC recommended receiver practice (**Ref A1-8**), but this curve can change in time with future DTV equalizer performance improvements.

The equalizer noise-enhancement experienced in the presence of *severe* multipath should not degrade the DTV receiver's 15-dB white noise threshold by more than a reasonable amount (e.g., less than 7 or 8 dB). Finally, for small amplitude echoes, the equalizer's cancellation range should be at least ±40 μsecs.

These equalizer performance requirements are also necessary for normal, non-DTS *indoor* DTV reception in urban areas where no line-of-sight (i.e., Rayleigh propagation) conditions are often experienced due to either nearby buildings or just the lack of a window facing towards the transmitter. These conditions can lead to 0 dB echo amplitudes, where the signals can vary with time and even change from post-echoes to pre-echoes as mentioned above. While the design goals stated above were not met by early generations of DTV receivers in the late 1990s, they have already essentially been met by most 5th generation (**5G**) receivers, and certainly 6th generation (**6G**) receivers. Nevertheless, these DTV receiver performance levels still limit the DTxN design in terms of acceptable echo amplitudes and delays that are allowed in the overlapping coverage regions. These limitations, in turn, affect the individual DTx cell sizes that are allowed and thus determine the number of required DTxTs to cover a given area. However, as DTV receivers continue to improve and expand multipath cancellation capabilities, DTxN design limitations will be reduced and DTx network design will continue to get easier. In the mean time, careful DTS design is required in order to keep the self-induced multipath within the acceptable cancellation range (both amplitude and delay) of consumer DTV receivers.

For an optimized DTS design, it is desirable to transmit a repeated *pristine* RF signal (i.e., with minimal signal distortion) from each DTxT. To do this, *digital* regeneration is required in DTx systems. This means that both ATSC <u>data</u> and <u>signal</u> processing must be performed locally within each transmitter rather than just passing an amplified and filtered RF signal with all its inherent circuit and propagation distortions. However, just the *time delays* involved with VSB demodulation, equalization, trellis-decoding, de-interleaving, Reed-Solomon (**RS**) decoding, and de-randomization alone would be in excess of the equalization cancellation range of typical consumer DTV receivers located in the overlapping coverage regions if the DTxTs used an over-the-air input signal. When the processing delays in the 8-VSB encoding and re-modulation process of the digital regenerator are also considered, these digital regenerator delays are even longer, and make matters worse. Again, this is the reason that most on-channel repeaters are *not* acceptable for DTV operation. Rather, the data input to the digital exciter must be conveyed to each DTxT site by some other means than an over-the-air, on-channel RF signal in order to allow for proper synchronization and timing (delay or advancement) of the entire DTx system.

DTx network design includes consideration of DTxT locations and radiated power, cell sizes, transmitter antenna patterns, frequency offsets, amplitude differentials (e.g., Carrier-to-Interference ratios, or **C/I**), and timing delay differentials. Terrain shielding often makes the DTx system design easier since it reduces the number of overlapping coverage areas, and the isolation that it causes is often the reason for needing DTx systems in the first place. System optimization is always desired when designing these DTx systems. Of course, it is likely that there will be some overlapping RF coverage regions, and some of them may prove challenging for certain DTV receivers (particularly their equalizers). Viewers can use highly directional receive antennas to help mitigate the problem, but this is not always a possible solution. Certainly, DTx network design can also be optimized by placing these most difficult overlapping regions in areas with little or no population.

Flat terrain areas sometimes require DTx systems (e.g., lack of tall transmitter tower space), and they are obviously more challenging as there is no terrain shielding to reduce overlapping coverage regions. The DTS designer must work hard to limit desired-to-undesired (**D/U**) ratios in overlap regions as well as keeping the self-induced echo delays in critical overlapping coverage regions within reason. RF signals, in free space, travel one mile in about 5.4 μsecs, so echo delays change twice as fast as this (i.e., 10.4 μsecs/mile) as the reception area moves away from one DTx transmitter and towards another one. Design goals are to minimize the D/U ratios and echo delays in overlapping regions with significant population. Obviously, these goals do not need to be met in the middle of lakes or in wilderness areas where few humans will watch DTV signals. In many cases, directional transmitting antennas will help alleviate some of the overlapping coverage area problems, and they will be further aided if these areas (e.g., rural areas) typically have a significant number of viewers using directional

outdoor antennas. These considerations, as well as the delay spread in the current generation of DTV receivers, will determine the cell size of an optimized DTx network.

As DTx networks become active and more commonplace, means must exist to easily measure and adjust the amplitudes and delays of the various transmitters at any field site installation. This system verification is critical, and therefore test methods and test equipment are required for field technicians to do this work. Preferably, this should be done while the DTx transmitters are all in service, without the need to shut them down individually. There is a method, however, to meet this requirement by use of an RF watermark signal that can be buried well below the DTV RF signal to avoid degradation of DTV receiver performance, yet allow accurate amplitude and delay measurements in the field (using power correlation methods to reduce the effects of the DTV signal) for both DTS network adjustment and verification. This topic is covered in a later section.

To summarize, the following are important considerations for DTxN design: DTxT site locations, DTxT antenna azimuth and elevation patterns, DTxT ERP, DTV receiver performance (e.g., VSB decoder IC generation), the physical location of a majority of people with DTV receivers, availability of outdoor receiving directional antennas, local terrain, and cell size. While early generations of DTV receivers, such as 1G through 3G (and even some 4G) had difficulties with large amplitude, self-induced DTS echoes, the excellent 5G (and later) designs have now permeated consumers' homes and allow reasonable DTxN performance. However, as larger percentages of deployed consumer DTV receivers in homes become 5G and later, DTS will play a larger role in many service areas that need additional help to be successful DTV regions.

Distributed Transmitter Synchronization

An important issue with a DTx design is the *requirement* for several VSB transmission parameters to be *synchronized* in all the DTx slave transmitters for the sake of DTV reception in coverage areas where the main and gap filler signals overlap one another. It is vital that the *exact* same symbol sequence (including the Data Field Synchronization (**DFS**) segments made up of both data field and data frame syncs) at the *exact* same symbol rate is modulated onto the *exact* same RF carrier frequency, and that these signals are transmitted from each transmitter at *very nearly* the same time. However, when two exciters are fed from the same SMPTE 310M MPEG transport data source, they will almost always generate different symbol sequences due to the *non*-deterministic (both repetitive and stochastic) ATSC *data* processing that relies on the exact location of the data field syncs with respect to the transport data packets as well as the initial conditions of the modulator's trellis-coding and pre-coding as well as its past data bit history. [*A stochastic process is one whose behavior is non-deterministic in that a state does not fully determine its next state, and is characterized by randomness - Wikipedia*]. Note that the ATSC *signal* processing that occurs in all VSB encoders does not alter the symbol *sequence*, but rather just alters signal characteristics like sync insertion, pilot insertion, root-raised cosine filtering, VSB modulation, and upconversion to the final RF channel. As long as the 10.762 Msymbol/sec, 8-level output of the ATSC data processor in each DTx transmitter is identical in sync timing and symbol sequence, the final RF outputs will all be identical as required.

When two DTx transmitters are *not* synchronized, they behave as co-channel DTV-into-DTV interferors rather than as echoes (i.e., replicas) of each other. Under this non-synchronized multiple transmitter condition, any DTV receiver in the overlapping coverage region can only have error-free reception with a D/U ratio of about 15 dB or more. However, if proper synchronization is accomplished so that each transmitter output has an identical symbol sequence modulated on the same frequency-locked RF carriers, D/U ratios near or at 0 dB (i.e., near or at 100% *echo* amplitudes) can be handled by the receiver if the time difference between the two signals is relatively small and well within the receiver's delay cancellation range. Therefore, synchronized distributed transmission systems potentially have a significant (about 15 dB) advantage over non-synchronized transmitter systems.

There are several issues to consider when discussing synchronization of an ATSC digital transmission signal. They are:

1) RF (pilot) carrier *frequency*
2) Data (symbol) clock *frequency*
3) *Symbol sequence*, which includes the exact MPEG transport data, the data field and data frame syncs locations with respect to the transport stream, and the special binary field sync bits that can carry mode of operation information
4) Relative *timing* of all the radiated signals

All of these factors play an important part in transmitter synchronization. *Frequency* and *timing* are fairly straightforward parameters to synchronize as well as the insertion of the field sync control data. The synchronization of an exact symbol sequence is a bit more complicated since normal ATSC data processing operations "free run," which means that they are not guaranteed to be identical in multiple transmitters unless some careful synchronization means is undertaken.

To provide further details, there are two primary areas of *non*-deterministic data processing in 8-VSB exciters, both of which are covered in the ATSC A/110B Synchronization Standard for Distributed Transmission (**Ref A1-4**).

First, some of the VSB data processing methodology, as shown in **Figure A1-3**, is ***repetitive***, occurring at fixed intervals and synchronized to the VSB data frame and field syncs. Those data processing components synchronized to VSB *field* syncs are data byte randomization with a 16-bit maximum-length pseudo-random binary sequence, the 52-segment convolutional byte interleaver, and the 12-symbol intra-segment bit interleavers. All of these processes are *initialized* in the VSB modulator at the beginning of each VSB data field before any processing of the data begins, but without any pre-determined relationship to specific points in the incoming MPEG data stream. This is due to the fact that there are no data field syncs contained in the normal ATSC MPEG stream that is input to the VSB modulator. Rather, a standard VSB modulator *arbitrarily* selects a spot to insert the VSB frame sync in the data stream. Therefore, while normal ATSC data processing essentially "free runs", it is imperative that *each* DTx transmitter inserts the data field sync at the *same* point in the MPEG data stream. Otherwise, a different bit pattern and consequently a different symbol sequence will occur among the DTxTs. Since the data frame occurs every second data field with its non-inverted middle-63 pseudo-random binary sequence (**PRBS**), only the data *frame* sync needs to be synchronized among the various DTxTs while the data *field* sync can be easily derived from it.

Second, part of the VSB data processing methodology, as shown in **Figure A1-3**, is VSB trellis-coded modulation, which includes trellis-coding of half the bits and pre-coding of the other half. It is a ***stochastic*** (non-deterministic) process whose result is dependent on the initial startup conditions as well as all the past data bits that have been processed. This process has no *fixed* relationship at all to the input data stream. The trellis-coding consists of a 4-state, linear Ungerboeck code (2 memory bits that together can have four different starting conditions) that is used to lower the error threshold in the receiver by means of soft Viterbi decoding. The pre-coding consists of a 2-state feedback process (1 memory bit that can have two different starting conditions) that is used to compensate for a comb filter in the receiver that reduces NTSC co-channel interference. The initial condition of these three bits affect the final bit (and ultimately symbol) sequence. To make matters more difficult, there is not just one trellis coder in the 8-VSB transmission system but rather 12 *independent* trellis coders in parallel that provide short symbol interleaving to complement the 12-symbol subtractive feed-forward comb filter in the receiver that reduces any NTSC co-channel interference that is present. None of the above memory bits is ever initialized or reset as part of the normal "free running" VSB data modulation process. That is, while the selection of *which* of the 12 parallel trellis/pre-coders is related to field sync (78 groups of 4 data segments is the trellis selection repetition rate that fits nicely into one 312-segment field), the trellis/pre-coding itself (*within* each of the 12 trellis/pre-coders) is *not* related to field sync at all or to any other element within the MPEG data stream since this coding process is "carried over" both data segment syncs and field syncs. Therefore this stochastic coding process must be synchronized in all DTxTs. This required data field sync carryover process for trellis/pre-coder processing is the reason that the last 12 symbols in the "binary" frame and field syncs are actually 8-levels, since they are a carry-over (i.e., a repeat) of the last 12 symbols in the last data segment that occurred just before the data frame and field syncs began.

This means that a total of 36 bits (i.e., 3 trellis/pre-coder bits times 12 trellis coders) have an arbitrary initial condition as does the exact placement of *frame* sync within any one of two groups of 312 data packets that occur between successive frame syncs. The odds of two transmitters, even with the required identical MPEG data stream at their inputs, having the *exact* same VSB symbol pattern at their outputs is 1 in 42 trillion (**Ref A1-7**). This means that there are 42 trillion different possible VSB symbol sequences for the same MPEG input data stream. Therefore, distributed transmission requires a means of synchronizing the *initial* VSB data processing conditions in the modulator and maintaining this synchronization over time.

Note that the effects of non-deterministic data processing or precise signal frequency and timing does not matter if a *single* transmitter is radiating a VSB signal to all DTV receivers in a given service area since any and all of the DTV receivers can decode this signal properly and provide valid pictures and sound. However, when *multiple* transmitters are required to be precisely synchronized to optimize DTV receiver performance in overlapping coverage areas, the non-deterministic data processing requires a method of synchronization that will make it deterministic for all slave transmitters.

Therefore, the following *four* items require consideration within a DTx system:

#1 The same VSB **carrier frequencies** of all distributed transmitters must be locked together so that the multiple DTV signals that are present at the input to any DTV receiver located within overlapping coverage areas will appear as *static* multipath rather than *dynamic* multipath, which is easier for the adaptive equalizer in a DTV receiver to handle.

#2 The same VSB **symbol clock frequency** must be used for all the transmitters in a DTx network in order to keep all the symbol sequences identical over time. In theory, if the ATSC standard is followed precisely, identical SMPTE 310M transport streams present at all exciter inputs would create identical symbol clock *frequencies* at the exciter outputs since the ATSC A/53E standard (**Ref A1-1**) requires fractional frequency locking of the VSB symbol clock and transport data stream clock. However, this ATSC system requirement is NOT followed strictly in many of

the deployed single DTV transmitter systems (i.e., without distributed transmission), and fancy data processing is often performed that involves adding or subtracting (if present) null packets from the MPEG stream and then re-stamping the PCR bytes. However, the danger exists that if there are not enough of existing null packets to delete from the MPEG stream (e.g., in the case of ultra-efficient opportunistic data or statistical multiplexing situations), loss of critical data can occur.

#3 The exact same **symbol modulation** must be transmitted from *each* transmitter. This requires not only the same MPEG data available at each of the modulator inputs, but also (a) the same placement of the VSB *frame sync* within the MPEG data stream, (b) the same field sync *control data*, and (c) and the same synchronization (initialization and tracking) of the *trellis-coders and pre-coders*.

#4 The **relative delay** between the various DTx network transmitter RF signals must be set so that a vast majority of receive locations in the overlapping coverage areas will *not* have large-amplitude DTS-induced echoes with *delays* greater than the echo cancellation range found in typical consumer DTV receivers. The time window that the earliest and the latest arriving echo signals fall within is called the delay spread (**DS**). For strong-signal multipath echo amplitudes greater than 18% (i.e., > -15 dB), the VSB data eyes will close and transmission errors will occur unless these correlated echoes fall within the DTV receiver's echo delay cancellation range.

The final DTx network design aspect is selecting the proper **radiated power value** and **antenna radiation pattern** for each of the transmitters so that there is enough signal strength for DTV reception in the desired areas but not so much that interference into nearby analog or digital signals is caused. This is a very delicate balance that complicates the DTS design.

Distributed Transmission System Implementation Techniques

Figure A1-4a illustrates a generic distributed *transmission* system that is based on the ATSC A/110B standard (**Ref A1-4**). Note that the "new" types of hardware that are required beyond that needed for normal DTV transmission are (1) the distributed transmission *adapter* (**DTxA**) and (2) distributed transmission slave *transmitters* (**DTxTs**). Understanding the operation of these new types of hardware is critical in understanding how the DTx system operates and how it should be optimized in the field. As a comparison, **Figure A1-4b** illustrates a generic distributed *translator* system, which has some additional system hardware requirements.

In order to properly synchronize the entire system, all the DTx transmitters are locked with respect to **RF carrier** and **symbol clock** *frequencies* (requirements #1 and #2 listed in the last section). From **FigureA1-4a**, it can be seen that the DTx system typically includes a Global Positioning System (**GPS**) satellite receiver for the DTxA as well as all the DTxT devices (LORAN-C can also be used as well). The purpose of the GPS receivers, which use reference signals from orbiting satellites, is to accurately provide both a common *frequency* (10 MHz) and a common *time* (1 pulse/second or 1 pps) reference to the DTxA and *each* slave transmitter.

The 10 MHz reference is used to generate precise IF modulator and RF upconverter local oscillator signals that will ultimately create precision RF carrier frequencies. The main source of frequency error is typically the IF modulator since the RF upconverter often will include straightforward phase-locked loop (PLL) synthesizers that can be locked to an external GPS 10 MHz reference source. An IF modulator that uses older, more traditional methods of *analog* IF modulation can do the same thing since the IF frequency is independent of the symbol clock. However, in modern VSB modulators that are often implemented digitally, the DSP clock is often locked to a multiple of the symbol clock frequency (e.g., x8), which is derived from the transport clock frequency (e.g., with a 313/564 ratio). The DSP clock is then used to create an IF signal. Therefore, transport clock frequency (and jitter) errors *may* affect the IF signal output frequency. However, IF frequency errors can be mitigated by carefully applying some form of frequency correction, thus making the IF frequency *independent* of the SMPTE 310M clock. One method is to make use of an accurate GPS 10 MHz reference for determining frequency errors in order to generate a correction. This frequency correction often is accomplished with low-frequency averaging methods that exhibits a correction lag time that may allow a small frequency error to occur if the SMPTE 310M clock frequency is slewing (**Ref A1-7**).

In addition to utilizing the GPS 10 MHz reference, the DTxA can provide some "frequency *smoothing*" to the SMPTE 310M signal through use of a *very* low bandwidth (i.e., "*very* slow") PLL. This allows different modulator designs to be used, and still maintain excellent DTxN performance. A desired goal is to keep the frequency error to less than 0.5 Hz of its nominal frequency so that all the DTxTs are within 1 Hz of each other, which minimizes the *apparent* Doppler shift among all of the DTx transmitter "echoes" that DTV receivers must handle.

The 1 pps GPS reference provides a precise time reference for accurately *determining* time delays and *creating* delay offsets. Common precision timing references at each transmitter are needed in order to accurately set the various timing delays from each DTxT so any existing multipath in the overlap coverage regions is well within the typical consumer DTV receiver echo delay cancellation range. When timing delays are compared to 1 second intervals, the 1 pps signal can be used in conjunction with the synchronous 10 MHz reference to create a 1-second tick signal for the DTxA and the DTxT units.

Therefore, both the GPS 10 MHz reference and the 1 pps timing pulses help with synchronization requirements #1, #2, and #4 listed in the previous section.

The ATSC A/110B standard also has *two* symbol frequency stability modes. In the distributed transmission packet (DTxP) that is used to carry timing and control information from the DTxA to the slave DTxTs, if the Stream_Locked_flag in the distributed transmission packet (**DTxP**) is set to a value of 1, the slave transmitters lock their symbol clock to the incoming transport stream to achieve compliance to the ATSC standard (**Ref A1-1**). However, the service multiplexer, which determines the transport stream frequency, should be accurate to within the recommended ± 2.8 ppm (**Ref A1-9**). While this is not be a difficult task since the ± 2.8 ppm frequency tolerance requirement is the same value used in the NTSC color television system from 1953, the service multiplexer can also use the GPS 10 MHz reference to create its output transport data stream, thus allowing all the downstream processing (DTxA and all the DTS slave transmitters) to be synchronized and therefore identical in their signal processing. Alternatively, if Stream_Locked_flag is set to a value of 0, then the GPS frequency reference system employed at all the transmitters is used to create the accurate data clock frequency. However, if the station's service multiplexer is not locked to the same external frequency reference (meaning that there is slight frequency difference between the service multiplexer and the DTxA transport clock rates), there is the possibility that null packets will have to be added to or subtracted by the DTxA from the transport stream (assuming there are enough of them available). The processing within the DTxA guarantees the output frequency stability to be within the required ATSC-recommended value of ± 2.8 ppm. More importantly, use of GPS locking of the transport stream data rate and of the slave transmitter data clocking becomes *necessary* when there are buffers in any of the STL paths that can result in reestablishing the data clock frequency. Even though buffer clocks might be within the SMPTE 301M standard permitted error, very large time delay differences can accumulate between transmitters from very small frequency differences occurring over a long time period.

The remaining synchronization tasks are initiated by the DTxA by processing the incoming MPEG transport stream from the encoder and inserting synchronization data before it passes the stream on to all the transmitters in the DTx system. The DTxA performs *four* functions in DTS synchronization:

1) Cadence sync insertion (for *frame* sync placement)
2) Side channel insertion (for *field* sync control data insertion)
3) ATSC channel coding (for synchronized data processing)
4) Distributed transmission packet insertion (to convey synchronization information to all transmitters)

In order to properly synchronize the data sequence in all the slave transmitters, some means is necessary to convey this synchronization or initialization information. Three methods are discussed below, and illustrated in **Figure A1-5**.

First, an IF version of the completely-modulated VSB signal can be conveyed to each DTS transmitter over an *analog* STL. Since no further data processing is required at the transmitter sites, all the transmitters will receive, and then pass along after RF upconversion to the same channel frequency, the exact same symbol stream. However, this requires a reasonably clean link (i.e., little noise or distortion) to all the transmitter sites since no further data processing (equalization or error-correction) is performed. It also requires GPS signals at all locations to create precise RF signals (within ±0.5 Hz) for quasi-static DTS-induced multipath in overlapping coverage areas as well as analog delay lines to adjust the relative system timing to minimize echo delays in these same areas. This method does *not* allow for exciter pre-correction to compensate for transmitter linear and nonlinear distortion.

Second, a *baseband* version of the completely-coded 8-level data stream can be conveyed to each DTS transmitter over an *analog* STL. Since no further data processing is required at the transmitter sites, all the transmitters will receive, and then pass along after modulation and RF upconversion to the same channel frequency, the exact same symbol stream. However, this again requires a reasonably clean link (i.e., little noise or distortion) to all the transmitter sites since no further data processing is performed. It also requires GPS signals at all locations to create precise IF and RF signals (within ±0.5 Hz) for quasi-static DTS-induced multipath as well as analog delay lines to adjust the relative system timing to minimize echo delays in the overlapping coverage areas. It also does *not* permit the use of exciter pre-correction of transmitter linear and nonlinear distortion. A similar method is to take this same signal, convert it into 3-bits for every symbol (2 data and 1 trellis), and then send it over a 32 Mbps STL (1.5 times the 21.5 Mbps VSB data rate) to all the transmitters, where it will then be converted back (with a digital-to-analog converter) into the same 8 levels for VSB modulation and RF upconversion. In this modified version, a digital delay line could be employed to set the delays for adjusting the relative system timing, and pre-correction of linear and transmitter nonlinear distortion would be possible.

Third, a normal MPEG-2 transport data stream at its typical 19.393 Mbps rate carrying some extra synchronization data within it can be conveyed to each DTS transmitter over a digital STL. Only a very small amount of the transport data bandwidth is used to convey this synchronization, timing, and control information to each DTS

transmitter, providing compatibility with the existing digital STL infrastructure and yet providing significant amounts of flexibility and control. It is this third method that the ATSC A/110B standard (**Ref A1-4**) describes.

The ATSC A/110B standard calls for ***three* mechanisms** to be added to the MPEG-2 transport stream that simultaneously transfers the appropriate synchronization information to all the transmitters, using the existing MPEG transport data stream that is sent from the service multiplexer to all the transmitters via the **SMPTE 310M** synchronous serial interface (**SSI**) standard that was designed specifically for DTV studio-to-transmitter links. However, no extra data bandwidth or error correction was added to this SMPTE 310M standard for this purpose. Therefore, to provide remote transmitters with the pre-coder and trellis synchronization, a very small amount of transport stream bandwidth must be allocated specifically for this purpose.

The **first** information transfer method is for the DTxA to insert a ***cadence sync*** (**CS**) signal into the MPEG transport data stream. Its purpose is to provide additional structure to the data stream that permits locking the *repetitive* transmission data processes (everything *except* trellis coding and pre-coding) in the DTxA and the DTx transmitters. The only purpose for this cadence sync insertion process is to remove the normally *arbitrary* selection of VSB data *frame* sync placement in a VSB signal by a transmitter (dependent upon when the modulator is first powered up). Therefore, the cadence sync defines a specific location within the transport stream for the insertion of the frame sync (requirement #3a listed in the last section) in all the DTxTs. Otherwise, all the transmitters would have arbitrary insertion of the VSB frame and field syncs into a VSB data stream, thus not providing identical symbol streams at a transmitter output.

Within the VSB transmission signal, there are binary data frame and data field syncs that are made up of pseudo-random *binary* sequences (511-bit PRBS and three repeated 63-bit PRBS) that can be used in DTV receivers to find the *location* of the data frame and data field syncs as well as serve as training signals for equalizers. However, every *other* data field sync has the middle 63-bit PRBS bit-inverted, which inherently creates a Data Frame structure composed of two Data Fields. Common ATSC nomenclature refers to the field sync with no inversion as the Data Frame Sync and the field sync with inversion as the Data Field Sync. Of course, there is a simple integer phase relationship between these two syncs: two Data Fields, *each* consisting of 313 data segments, occur in one Data Frame that has a total of 626 data segments (*inclusive* of the binary field and frame sync data segments). Therefore, if the Data Frame were synchronized in the VSB modulators, the Data Field sync would be automatically synchronized as well.

MPEG transport data streams *always* have a 47hex MPEG sync byte at the beginning of each packet, allowing proper synchronization within MPEG decoders due to its fixed 188-byte repetition rate. However, this DTx synchronization method periodically inserts a cadence sync by replacing this 47h sync byte with its *inverted* bitwise code (i.e., B8h) every 624 data packets so that it is easily recoverable. The reason that cadence sync is inserted every 624 MPEG packets (rather than every 626 or twice 313 as in the VSB signal) is due to the fact that no frame or field syncs exist in the **MPEG transport** world, only in the **VSB world**. Therefore, the DTxTs must insert a frame or field sync segment packet every 312 MPEG data packets, which means that in the DTS transmitters the VSB field *sync* repeats every 313 data segments and the VSB *frame* sync repeats every 626 data segments after the data stream is expanded by this sync insertion process.

The detection of a cadence sync (i.e., inverted MPEG sync byte) causes a slave transmitter to insert a Data *Frame* sync into the data stream that is defined as having ***no*** inversion of the middle PN-63 sequence (as opposed to the Data *Field* Sync which ***does*** have inversion of the middle PN-63 sequence). In DTxTs transmitters, a Data Frame sync is inserted into the stream immediately ***before*** the RF emission of the MPEG transport packet that contains the cadence sync. All MPEG sync bytes and cadence sync bytes are removed by the DTS transmitters and replaced with the 4-symbol (equivalent to one byte) VSB binary data segment sync before transmission, as expected by consumer DTV receivers. Note that, in the master timing control unit called the DTxA, the data frame cadence sync has arbitrary phase with respect to the MPEG transport data stream, but in all the DTx transmitters, it does *not* have arbitrary phase but rather it has the exact same relationship to the MPEG transport stream that exists in the DTxA. In other words, the DTxA synchronizes all the DTx transmitters via the cadence sync.

In addition to the cadence sync, the position of the Data Frame Sync can alternatively be determined from a 10-bit Packet Number that is part of the DTxP, as explained below. This 10-bit number designates the number of MPEG data packets that have passed since the last cadence sync (which represents Data Frame Sync), including the DTxP itself. The maximum value of the packet number is 624, which is covered by a 10-bit binary number (which has 1023 as its maximum value). The cadence sync is used in most DTx systems, but when distributed ***translators*** are employed that have multiple tiers (i.e., multiple cascaded levels), the cadence sync can *not* be transmitted through the RF system (since Data Segment Sync is used in the VSB domain instead of the MPEG 47hex sync). It would take a special VSB receiver to reinsert a cadence sync (which is certainly a possibility), but the redundant transmission of the Packet Number alleviates the need for a special VSB receiver.

With either approach, this first information transfer method allows for the ATSC data processing blocks to be synchronized to the VSB's frame sync.

The **second** information transfer method is for the DTxA to convey ***field sync control data*** for insertion into the VSB binary field sync at each DTx transmitter (requirement #3b listed in the last section). Besides the 24 bits that convey the VSB transmission *mode* (e.g., 2-VSB, 4-VSB, 8-VSB, 16-VSB, 8T-VSB), there are an additional 92 bits that immediately follow that are reserved for future use (e.g., future transmission modes). For example, some of these reserved information bits can be used for E-VSB which requires the dynamic robust channel map to be conveyed to any E-VSB transmitters once per field. Therefore, in addition to providing *control* of the transmitter mode in a DTx system, some of these DTxP bits are also used to *inform* DTV receivers out in the field which specific data mode that is being transmitted. Any changes in the actual transmitter mode starts upon the end of the first field *after* the bits are received (**Ref A1-4**). By transmitting all of the possible transmission mode bits (current and future) from the DTxA to all the slave transmitters, the possibility exists for additional processing to be performed at the MPEG source end that can add the necessary information bits to the DTxA for inclusion with the new enhanced data stream so that the same DTx transmitters can be used for a new service as well as for DTx networks. However, the mode data is *not* sent in the DTxP with the other data, but rather is conveyed in a *side channel* so that it can be updated quickly at a field rate rather than only occasionally at a slow DTxP rate (e.g., once per second).

This particular *Transmitter Mode Control Data* is formatted into 19 data bytes followed by 20 Reed-Solomon forward error correction bytes (total of 39 bytes, or 312 bits), which is sent using the MPEG transport data stream. The ATSC A/110B standard (**Ref A1-4**) refers to this path as the *Field Rate Side Channel*, and it is created by using the single MPEG "transport_error_indicator" bit within *each* individual MPEG packet. Therefore, 312 transmitter mode control bits are sent serially ***every*** field since there are 312 MPEG transport data packets transmitted within one VSB data field (exclusive of VSB frame or field sync). Two fields worth of side channel data occur in between two cadence signals. This control data is sent out constantly, and *not* in a bursted mode, and can update the VSB control data every VSB field, if needed. Note that *no* precious payload data is used in this method since the MPEG packet error indicator is not typically utilized in this particular path; however, this error indicator bit in each MPEG data packet is *restored* to its proper value of zero (indicating no error) in each of the DTS transmitters just prior to transmission over the air. This amount of data is enough to send the 24 VSB mode bits and the 92 reserved bits that are inserted into each binary VSB field sync, and still leave an additional 36 bits in reserve for some other future transmitter mode control use (although there is no more room to send any additional bits in the binary VSB field sync). However, it is critical that this side channel data path be very reliable, which is why the 19 data bytes are heavily protected by 20 Reed-Solomon error-correction bytes.

When either the VSB mode state or the DFS reserved bits change, the DTx transmitter inserts the bits into the *next* VSB data field or data frame sync that occurs. However, any transmitter *mode* or *characteristic* changes are delayed for one field after that and begin upon the occurrence of the data segment sync (first byte) of the *subsequent* data field sync.

If, by chance, there is no input to the DTxA, the DTxA creates a default Field Rate Side Channel using the MPEG "transport_error_indicator" bits in all of the null packets that it internally creates for transmission. The default pattern that it transmits for the Field Rate Side Channel is identical with those bits nominally specified by the ATSC A/53E standard (**Ref A1-4**).

The side channel is currently useful for E-VSB system operation that sends some of the packets with various levels of increased robustness over 8-VSB packets but with reduced data rate (½ -rate and ¼-rate). E-VSB operation *allows* for the possibility of changing E-VSB field sync data bits once per field, if required. The distributed transmission data packet described in the next paragraph can not be used to send this data since its repetition rate is too low to send this VSB field rate data. It is important to note that the DTx system synchronization standard was created in such a way that a distributed transmission slave transmitter is *already* E-VSB compatible. The only required change is the DTxA, which would require a new unit to properly combine the multiple data-rate streams before performing the required DTxA processing. This technique can easily be extended to *future* VSB-related transmission systems, such as the ones being developed for mobile and hand-held technology that also need to carefully place certain data packets at certain locations within the VSB data frame. **Table A-1** summarizes this transmitter mode control data format.

Table A1-1 Field Rate Side Channel Data Organization

Definition	# of Bits	Comments
VSB Mode Data Setting	24	Definition of 8T-VSB and 16-VSB Modes; **inserted** into field sync (see A/53E)
DTS Reserved Data	92	Reserved; Some bits are used for E-VSB; **inserted** into field sync (see A/53E)
Reserved	36	For future use; ***not*** for insertion into field sync (no room left in VSB frame sync)
Side Channel ECC	160	20 bytes of Reed-Solomon error correcting code; Protects 19 byes of Side Channel Data; ***not*** for insertion into field sync
TOTAL	312	Same number of data segments in a VSB field (exclusive of frame or field sync)

The **third** information transfer method is for the DTxA to synchronize all the slave transmitters with regard to ***data symbol emission*** and ***relative delay*** (requirements #3c and #4 listed in the last section). The data symbol emission synchronization process encompasses data randomization, Reed-Solomon (**RS**) encoding (t=10, 187, 207), convolutional byte interleaving (B=52, m=4, N=208, BxM=N), and trellis-coded/pre-coded processing complete with 12-symbol interleaving). Some of this *data* processing is related to *repetitive* processing that is synchronized to the VSB field sync and some of it is related to the *stochastic* process that is dependent on the modulator's initial conditions and past data history. This is the most sophisticated processing that the DTxA unit performs since it is a model of the *exact* same data processing that is performed in every VSB encoder/modulator as part of the normal ATSC transmission encoding (**Ref A1-1**).

However, to maintain data *symbol* emission synchronism in a DTx system, every transmitter within the network must perform its data processing on the *exact* same MPEG transport data stream that the DTxA unit uses, complete with proper identification of frame sync placement within the transport data packet stream. In order to achieve this goal, the DTxA must distribute synchronization information to all the DTxTs in a special synchronization packet (i.e., the DTxP) that allows "slaving" (i.e., synchronizing) the DTxTs transmitters to both the *repetitive* and *stochastic* processes described earlier. While the placement of the VSB frame and field syncs is accomplished with the cadence sync byte that synchronizes the repetitive processes in all the DTxTs transmitters, the remaining synchronization is accomplished by conveying the trellis/pre-coder stochastic *state* data to all slave transmitters and removing any initialization ambiguity.

The DTxA, which is the "brains" of the DTS synchronization process, is required to have an MPEG-transport stream input and output port as well as frequency (10 MHz) and timing (1 pps) reference input ports and transmitter control input ports (see **Figure A1-4a**). The transport data input port receives all the MPEG-2 transport packets from the DTV station's service multiplexer, which is the device that assembles (i.e., combines or multiplexes) the individual data packets from all the various video, audio, and data services. The most common interface in use today is the SMPTE 310M standard, although other interface standards can be used (e.g., Asynchronous Serial Interface, or **ASI**).

In order to guarantee meeting the requirement for identical data, the DTV station's service multiplexer must be programmed to add a reserved *placeholder* packet to the MPEG transport data stream with a special *Packet Identification* (**PID**) value of 0x1FFA at a pre-determined repetition rate that contains fixed (i.e., static) placeholder data. The pre-determined repetition rate is not critical, but is often selected by the DTS designer to be around 1 second. The special placeholder packet is ultimately modified in the DTxA unit by replacing the temporary data from the station's service multiplexer with actual synchronization data. The DTxA then conveys this synchronization information to all the slave transmitters. If the service multiplexer does not provide this special "pre-formatted" reserved packet, then the DTxA might possibly remove an active MPEG-2 transport data packet to insert the necessary synchronization data packet. This can possibly cause unacceptable video or audio errors if no null packets are available from the service multiplexer (e.g., if opportunistic data is being transmitted or statistical multiplexing is being performed that uses up all the available packets). This particular reserved packet is called the pre-processed *Distributed Transmission Packet* or **DTxP** and conforms to the MPEG-2 Transport Stream packet structure defined in ISO 13818-1 Systems (**Ref A1-4**). This means that this reserved packet has the usual 4-byte header that includes the 0x1FFA PID information and a 184-byte payload. These pre-formatted static placeholder packets in the service multiplexer will ultimately be converted into synchronization packets by the DTxA.

For synchronization purposes, the ATSC's A/110B standard (**Ref A1-4**) defines a specific form of the *Operations and Maintenance Packet* (**OMP**) structure that is part of the overall ATSC transport data structure. This OMP structure supports a variety of *operations* and *maintenance* functions in a DTV transmission system. The DTxP is the first such packet *type* defined in the ATSC's OMP structure, with other functions possibly to be defined in the future. The OMP contains a 184-byte payload, just like any other MPEG packet. However, there can be more than one *type* of OMP, which provides system flexibility. The various OMP types are defined by the first byte (**OM_type**) of the 184-byte payload, which for a DTxP can be any value between 0x00 (0 decimal) and 0x0F (15 decimal). Other values of OM_type are reserved for future standards. The value of the OM_type byte for DTxP use indicates the *tier* of a cascaded translator network to which the packet is addressed, where tiers are assigned in sequence proceeding *away* from the source and incrementing by one for each successive tier in the cascade (**Ref A1-4**). A maximum number of 16 different tiers are allowed in the ATSC standard (main

DTx system plus 15 levels of distributed translators). As shown in **Figure A1-4b**, multiple DTxAs can be cascaded to create multiple DTxPs in order to control multiple translator tiers within the network. Each group of DTxPs (one group for each translator tier) will be used only by the associated DTS transmitters in that tier since each group of DTxPs will have a unique OM_type byte. However, for normal DTx systems or distributed translators with *one* tier, this OM_type value is set to 0x00.

Therefore, not only must a fixed known data pattern be sent by the station multiplexer to the DTxA via the OMP (i.e., the *pre*-processed DTxP), this same fixed known pattern must also be *re-inserted* by every slave transmitter, replacing the trellis and pre-coding state synchronization data (as of the snapshot time) that was inserted at the *output* of the DTxA and transmitted on the STL. This is necessary since the trellis and pre-coding state data that is conveyed in the DTxP is not part of the actual MPEG data to be transmitted is not part of the data that went through the trellis and pre-coder coding process. Rather, this special data is a temporary snapshot of the 36 trellis and pre-coding memory states that are created in the replica of the ATSC data processing model (**Figure A1-3**) affected by the actual MPEG data. In other words, the DTxA and all the DTxTs must perform VSB data processing on the exact same data (and *only* that data) that will be transmitted as an RF signal over the air. **Therefore, the 36 bits of trellis and pre-coding state synchronization data exists only on the STL, and are *not* transmitted over the air nor are they ever manipulated by the ATSC data processing circuitry. Therefore, this fixed known "stuffing" data pattern must be pre-determined and used by both the DTxA and the DTxTs.**

In summary, after the service multiplexer sends out the 4-byte MPEG header and the OMP packet type data, the remaining 183 bytes of the OMP payload (OM_payload) contains fixed placeholder data, most of which is initially filler that will ultimately be replaced by actual synchronization and control data in the DTxA, thus turning the OMP into a DTxP. This OMP filler data created in the service multiplexer, which is reserved for synchronization and control, is filled with alternating bytes *after* the first payload byte (which is the OM_type), starting with 0x55 in the second payload packet and every subsequent even-number packet, and then 0xAA in the third payload packet and every subsequent odd-numbered packet. This means that the 184-byte payload data has one byte of OM_type, 92 bytes of 0x55, and 91 bytes of 0xAA. The definition of this form of OMP (also known as a DTxP *precursor* packet) is shown in **Table A1-2** below:

Table A1-2 Station Service Multiplexer Output: DTxP Precursor Operations and Maintenance Packet (OMP)

Data	# of Bits	Comments
MPEG Sync	8	MPEG Header: 47h sync byte
Transport_Error_Indicator	1	MPEG-Header: 0h
Payload Unit Start Indicator	1	MPEG-Header: 1h
Transport Priority	1	MPEG-Header: 1h
Packet Identification (PID)	13	MPEG-Header: 0x1FFA
Transport Scrambling	2	MPEG-Header: 00h
Adaptation Field Control	2	MPEG-Header: 01h
Continuity Counter	4	MPEG-Header (increments every OMP packet; from 0000 to 1111)
Operations & Maintenance Packet Type	8	OMP Type: 0x00 (DTx rather than DTxR, which can be 0x00 – 0x1F)
Fixed Data Pattern #1	736	Payload: 0x55 (92 bytes; only even-numbered payload packets)
Fixed Data Pattern #2	728	Payload: 0xAA (91 bytes; only odd-numbered payload packets)
TOTAL	1504	188 bytes = 4-byte header + 1 Type byte + 183 Data bytes

The OMP *pre*-processing in the station multiplexer readies the system for creation and distribution of the DTxP by the DTxA. The DTxA, which controls the DTS synchronization process described above, typically receives the SMPTE 310M signal from the station's service multiplexer, performs a *two*-step process that creates a DTxP (as described below), and then sends an MPEG transport stream with the inserted DTxP synchronization data over a SMPTE 310M link to all the slave transmitters. The following information briefly describes how the DTxA accomplishes all of these tasks and meets all the necessary DTS synchronization requirements.

Synchronization of multiple ATSC transmitters requires the exact same transport data at the inputs to the data processing sections of their VSB encoders in order to achieve emission symbol matching. However, even with this condition met at all the DTS transmitter inputs, there is still no guarantee that emission symbol synchronization will occur unless the DTxA sends further synchronization information (trellis and pre-coding *states*) to lock up the non-deterministic *stochastic* transmitter processing (i.e.., trellis coding which provides a lower receiver threshold SNR value and pre-coding which aids the receiver's NTSC rejection-comb filtering). This trellis and pre-coded data is *not* synchronized with the VSB frame or field syncs, but is dependent on a VSB modulator's startup state initialization and all the subsequent data history.

From the discussion above, it is clear that the OMP must have *some* of its placeholder bits replaced with the actual DTS synchronization data to form a DTxP. The DTxP data format and definitions are shown in **Table A1-3** below. In the first part

of a two-part process, the timing and control data (to be described subsequently) that is inserted into the DTxP (replacing ***most*** of the placeholder bytes) is determined apart from the transport packet stream data itself. However, the following data is ***not*** inserted: (1) the trellis/pre-coder state data and (2) the Reed-Solomon FEC bytes that protect the 164 bytes of DTxP payload data. Since the trellis/pre-coder and Reed-Solomon information is dependent on the actual transport stream data sent from the service multiplexer, it is not known until *after* the OMP has been fully data processed within the DTxA just prior to sending it over the STL to all the transmitters. Therefore, *most* of the DTxP payload data is inserted into the OMP immediately upon its PID detection within the DTxA (prior to trellis and pre-coding processing). However, the special *stuffing* data bytes (0xAA and 0x55) which are temporary placeholders for the 12 bytes (7th through 18th) for trellis and pre-coder states as well as the 20 bytes (169th through 188th) for Reed-Solomon error correction code are left in the transport stream for *simulated* VSB data processing within the DTxA. For proper RF data symbol emission synchronization, these same stuffing data bytes must also be present in each of the DTS slave transmitters when the *actual* VSB data processing is subsequently repeated there just prior to transmission over the air.

This DTxP synchronization data (including the remaining 12 + 20 = 32 bytes of *stuffing* data) is processed in the DTxA through the *free-running* standard VSB data processing model (**Figure A1-3**) along with all the other transport stream data that has come before it. The data processing model establishes the various *timing relationships* (e.g., with respect to MPEG/segment sync and cadence/data frame sync) between the input transport stream and the several data processes that occur in a transmitter (**Ref A1-4**). This part of the DTxA processing can free run since the Distributed Transmission Adapter is the device that arbitrarily determines where frame sync will be placed and is the reference for all the transmitters.

Stochastic state synchronization is then accomplished by capturing a snapshot of the 36-bit trellis/pre-coder states (3 groups of 12 bits) at well-defined times that exist in the DTxA simulated VSB data processing model (i.e., at the end of *each* equivalent VSB data field or frame sync). The three groups represent the two trellis states and the one pre-coder state, and the 12 bits represent the twelve symbol intra-segment interleaver. Therefore, these 36 states represent the status of the trellis/pre-coder storage devices as they will exist immediately *following* the Data Field Sync at the start of the *next* Data Field after the appearance of the DTxP in the MPEG-2 transport stream (**Ref A1-4**), assuming that the proper stuffing data is appropriately inserted at the DTxTs. Before insertion into the DTxP, these 36 trellis/pre-coder state bits are extended to 96 bits by adding a 4th parity bit to each group of 3 bits and then appending to each 4-bit group an inverted second copy of the 4 bits for extra *robustness* for a total of 8 bits per trellis/pre-coder state (i.e., 8 bits x 12 groups = 96 bits). After the trellis and pre-coder states are known and inserted into the DTxP, the 20 Reed-Solomon error correction bytes are determined and then inserted at the end of the DTxP for transmission to the slave transmitters over the normal STL.

It is desired to send from the DTxA to the DTxTs the trellis/pre-coder states in a DTxP that occurs *before* the time these states actually exist in the trellis/pre-coders, thus allowing enough time for the DTxTs to properly process them and insert them accordingly into the final output stream. This requires the transport stream to be delayed by one data field in order to obey the physical laws of causality (in the real world). Therefore, the presence of a *data field delay* in the DTxA then allows for the state of the trellis/pre-coder at the *end* of a data field to be inserted into a DTxP that is transmitted *before* the end of that particular field. This facilitates the subsequent symbol synchronization in the slave transmitters.

Once again, it is very important to note that **all the DTxP data is passed through the VSB data processing algorithm *including* the 12 bytes of 0x55 and 0xAA stuffing data that are *placeholders* for the 96 trellis/pre-coding bits and the 160 Reed-Solomon bits (20 bytes), but *not* the actual trellis/pre-coding state data and Reed-Solomon data since they are a result of the data processing itself and are not known until *after* the data processing is complete at a well-defined snapshot time.**

In order to achieve DTS synchronization, the DTxP synchronization data is then sent to all the DTS transmitters. At each slave transmitter, the trellis/pre-coder state data and the Reed-Solomon error-correction data is immediately detected (robustly) and stripped from the DTxP and replaced with the exact same default filler data (multiple 0x55 and 0xAA bytes) that was present in the DTxA when the initial VSB data processing was performed. This step is critical in order for each slave transmitter to be able to produce an exact replica of the DTxA symbol sequence at its output and thus provide system synchronization for all the RF outputs. The remaining DTxP data is also stored in the slave transmitters for use in timing synchronization and other control functions, to be discussed below, but is also transmitted over the air with all the other MPEG transport stream data.

At the end of the next VSB field sync after a DTxP is detected in the DTx transmitters, each DTxT transmitter is slaved to the trellis/pre-coding states determined in the DTxA data processing model. After the first successful synchronization of all the slave transmitters has occurred, every subsequent processing of the DTxP data in the slave transmitters should track the DTxA data and only serve to *verify* synchronization. When the sync locations and the trellis/pre-coder states match, all the transmitters are running in "lock step." Only if there is an error in the transport stream data to a transmitter or if some unusual event (e.g. a "glitch") occurs in the operational timing of a slave transmitter will it be necessary to resynchronize a transmitter during normal operation. In this case, the DTxP data is actually used to "jam sync" the trellis/pre-coder memory storage

devices during the VSB frame and field syncs and bring the offending transmitters(s) back into lock just *prior* to the beginning of the first data segment following the VSB frame or field sync.

Remember that since the DTxP replacement control and timing data that is inserted into the transport stream passes through the DTxA's VSB data processing, all of the slave transmitters must likewise process this same data and then transmit it *over the air* in order to achieve matching trellis code trajectories and keep the transmitters synchronized. This control and timing data can be received remotely and used by special test and measurement equipment in the field. The only data that is NOT transmitted over the air is the trellis/pre-coder states and the DTxP Reed-Solomon error correction codes used in the slave transmitters since they were not processed by the DTxA data processing (but were actually a *result* of that processing).

Table A1-3 DTxA Output: Complete Distributed Transmission Packet (DTxP) Organization

Data	# of Bits	Comments
MPEG Sync	8	MPEG Header: 47h sync byte
Transport_Error_Indicator	1	MPEG Header: 0h (unless Tx mode control data is being sent)
Payload Unit Start Indicator	1	MPEG Header: 1h
Transport Priority	1	MPEG Header: 1h
Packet Identification (PID)	13	MPEG Header: 0x1FFA
Transport Scrambling	2	MPEG Header: 00h
Adaptation Field Control	2	MPEG Header: 01h (no adaptation field, just payload data only)
Continuity Counter	4	MPEG Header: (increments every OMP packet; from 0000 to 1111)
Operations & Maintenance Packet Type	8	OMP Type: 00h (DTxTs rather than DTxRs)
Reserved	8	Payload: Reserved for future DTxP use
Trellis Code State (3-bit pre/trellis coder)	96	Payload: (3-bits+1 parity) x 2 copies x 12 parallel coders; data sync-ing
Synchronization Time Stamp	24	Payload: For time delay adjustment & optimization
Maximum Delay	24	Payload: Maximum time delay in DTxN (DTxA Output to Tx Output)
Network Identifier Pattern	12	Payload: Network Watermarking ID
Stream Locked Flag	1	Payload: Tx symbol clock lock requirement (internal TS=1 or external GPS=0)
Reserved	1	Payload: Reserved for future DTxP use
Packet Number	10	Payload: Supplements cadence sync for VSB frame sync location
Reserved	32	Payload: Reserved for future DTxP use
Tx Group Number	8	Payload: 1 byte; 1st 8-bits of Tx addresses (for Watermarking ID, *redundant*)
Tx Address (12-bits each)	192	Payload: Transmitter Watermarking ID (for up to 16 TXs)
Tx ID Level (3 bits each)	48	Payload: Amplitude level of Watermarking ID (for up to 16 TXs)
Tx Data Inhibit (1 bit each)	16	Payload: Watermarking *modulation* ON/OFF control (for up to 16 TXs)
Tx Time Offset (16 bits each)	256	Payload: Emission timing adjustment (for up to 16 TXs)
Tx Power (12 bits each)	192	Payload: TPO amplitude control (for up to 16 TXs)
Reserved (4 bits each)	64	Payload: Reserved (for up to 16 TXs)
Reserved	320	Payload: Reserved for future DTxP use
DTxP Forward Error Correction	160	FEC: Reed-Solomon (t=10, 164, 184); protects all bytes except header
TOTAL	1504	**DTxP**: 188 bytes = 4-byte header + 1 Type byte + 163 Data bytes + 20 RS

Once the emission symbol sequence is synchronized among all the slave transmitters, the next critical characteristic to optimize is the relative *timing* of each transmitter with respect to the others. DTV receivers in overlapping coverage regions must deal with these self-induced DTS multipath echoes. The goal is to minimize the areas in the overlapping region that have large echo amplitudes with echo delays greater than what typical consumer DTV receivers can handle. In order to do this, a *common time reference*, such as provided by a GPS receiver, is needed at all transmitter locations as well as the DTxA site, preferably a reference that is accurate within 50 nsec at all locations. Once this absolute time reference has been established (e.g., the *leading* edge of the GPS 1 pps signal), then individual transmitter timing offsets can be determined and controlled remotely from one location through the DTxP signal from the Distributed Transmission Adapter. The use of GPS receiver technology at each site provides both an accurate frequency (10 MHz) and timing (1 pps) relationship.

Adjusting the proper timing among all the slave transmitters deals with *two* separate issues. The first is to determine all the various delays within the DTS *distribution* network itself, including the STL. The second is to decide where to place the best-case and worst-case relative signal timings (including RF propagation delays) within the coverage areas. However, this second issue is a matter of system design optimization and is not a part of the system timing control's logical definition described here.

There are three time references values in the DTS synchronization scheme that are distributed over the DTx network via the Distributed Transmission Packet. Two of these references values are sent to *all* transmitters: the *Synchronization Time Stamp* (**STS**) and the *Maximum Delay* (**MD**). The third time reference value is an *Offset Delay* (**OD**) that is *individually* sent to specific transmitters to bring them into line for optimal and acceptable relative emission timing.

The first timing variable that is included in the DTxP is the **Synchronization Time Stamp**, which is the number of 10 MHz GPS clock periods (i.e., 100 nsec time intervals) between the *last* leading edge of the 1 pps GPS clock tick and the time that the first DTxP bit is sent out from the DTxA. This STS value is determined from the 1 pps GPS timing reference at the DTxA and then sent to every DTS transmitter over the STL. Since every transmitter *must* also be outfitted with a GPS receiver, this provides an absolute time reference for the entire system. All timing will be described with respect to the STS reference value.

The second timing variable that is included in the DTxP is the **Maximum Delay** value, which must be greater than the approximate worst-case time delay from the DTxA output to the RF output of the *latest* transmitter. This value, which is measured in 100 nsec time intervals (i.e., 10 MHz reference clock periods), should be selected *by the DTx network designer* based on the sum of the delays of the longest DTx distribution link and the longest transmitter processing. Its purpose is to guarantee that the RF signal output times of all transmitters in the network have enough delay to account for (i.e., match) the longest delay in the distribution path to any slave transmitter input plus the delay of the transmitter and antenna. This time relationship is what provides for the possibility of relative *advancement* of a given transmitter emission time without violating the physical laws of causality. The MD value must be in a reasonable range so that when the individual slave transmitter Offset Delay values (which can be either positive for delay or negative for advancement) are added to it, the final value will be positive and represent a *delay* between 0 and 1 second. *Absolute* advancement, which is negative time, is impossible due to the physical laws of causality (i.e., in real-world limits). If the calculated value is greater than 1 second, then a value of 1 second is subtracted from the calculated value to accurately reference the emission time to the most recent 1 pps clock tick.

The third timing variable that is included in the DTxP is the **Offset Delay** value, which is a value that is individually selected for *each* transmitter to provide optimum timing performance of the DTx network. It is measured in 100 nsec time intervals (10 MHz reference clock periods) and can be either a positive or negative value (-32,768 to +32,767 which is -3.2768 msecs to +3.2767 msecs) that is compensated within each exciter by its Transmitter and Antenna Delay (**TAD**) value that is stored in the transmitter's exciter for subsequent *calculations* of precise emission time.

TAD is officially defined as "the time from the entry of the first bit of a Cadence Sync word into the Data Randomizer (first block of Data Processing system) to the appearance at the antenna output of the leading edge (zero crossing of the +5 to -5 transition) of the segment sync of the corresponding Data Frame Sync data segment" (**Ref A1-4**). This technique provides a *constant* value of TAD that is independent of all the other *variable* delay processing that occurs within the exciter's data processing (e.g., convolutional byte interleaving does not provide constant data delay as some bytes are spread out longer than others). Then, the exciter delay value plus the delays of all the RF components (high power amplifier, harmonic filter, emission mask filter, any channel combiners in the path, transmission line, and antenna) yield the final value of TAD. This value of TAD, which can be provided by the equipment manufacturers, is then stored in *each* transmitter where it is simply subtracted from the OD value that is sent in the DTxP by the distributed transmission adapter.

The reference time T_{REF} for synchronizing the emission sequence timing is the synchronization time stamp plus the maximum delay (i.e. $T_{REF} = STS + MD$). Every transmitter can *calculate* this *reference emission time* if (1) a working GPS receiver is present with an available 1 pps signal, and (2) a working transport data stream receiver within the exciter is properly extracting the MD value from the DTxP. Then the individual Offset Delay for each transmitter is added to this reference emission time, minus the individual TAD value that is stored in each DTS transmitter. That is, the desired Transmitter Emission Time ($\mathbf{T_E}$) is described by:

$$T_E = T_{REF} + OD - TAD = STS + MD + OD - TAD \qquad (1)$$

This emission time T_E is the desired time (relative to the last 1-second GPS clock tick) for *each* slave transmitter to begin radiating the VSB frame sync so that the required relative timing is met. However, once the desired emission time T_E is calculated for each transmitter, the amount of required transmitter delay (**Tx Delay**) must be determined by each transmitter in order for it to know how to adjust its internal variable data delay line so that the desired emission time is met.

Transmitter delay is achieved with a variable data delay line in each slave transmitter, which is the means for adjusting the timing of each transmitter's radiated output. It is defined as the delay between the leading edge of the arrival of the first bit of the MPEG sync byte of DTxP at the exciter input ($\mathbf{T_{IN}}$) to the DTxP first bit of the MPEG sync byte at the Data Randomizer ($\mathbf{T_R}$). The desired amount of variable transmitter delay ($\mathbf{T_D}$), in 100 nsec steps, is found by:

$$T_D = T_R - T_{IN} = T_E - TAD - T_{IN}$$

where T_{IN} is *measured* by the transmitter, in 100 nsec time intervals, the time from the leading edge of the last 1 second clock tick to the first bit of the MPEG sync byte of the DTxP.

The relative timing variables are illustrated in **Figure A1-6**, which is taken from the ATSC A/110B document (**Ref A1-4**). If the calculated emission time exceeds 1 second (which is the 1 pps clock tick rate), then 1 second is mathematically subtracted from the total emission time value (i.e., $T'_E = T_E - 1$) so that the new emission time is determined relative to the *most recent* 1-second tick time reference.

As an aside, note that the maximum delay must be properly selected to be greater than the sum of the transport delay, transmitter exciter delay, and the TAD. Also, in **Figure A1-6**, the DTxP Arrival time denotes the arrival of the first bit of the MPEG sync byte of the DTxP to the transmitter's external SMPTE 310M input connector while the DTxP Modulation time signifies the arrival of the first bit of the MPEG sync byte of the DTxP to the VSB data randomizer input contained within the VSB exciter. The difference between these two times is called the Tx Delay, and the sum of the Tx Delay and the TAD together make up the total transmitter system delay.

In addition to all the timing synchronization aspects, the ATSC A/110B standard also allows individual control of the transmitter power. Transmitter power level control may be needed to obtain the desired DTx network design, providing proper D/U ratios in the critical areas.

Distributed *translators* are very similar to generic distributed transmission systems. However, a major difference exists when multiple DTxR transmitters are cascaded more than one level (i.e., tier) deep and they are receiving their transport stream from another RF transmitter (i.e., over the air). When this situation exists, the 2nd tier and beyond typically does not have access to the trellis/pre-coder states from transmitters in earlier tiers since this information is only carried in the STL signal and is *not* transmitted over the air. The exact solution will depend on the complexity of the translator system design.

If the DTxR system is a simple *single*-hop implementation, where the main transmitter is on one frequency and all the distributed translators that receive this RF signal (which is acting like an STL) are to be synchronized on the same channel, but different from the main transmitter's channel, the solution is straightforward. In this case, the main transmitter is receiving the transport packet stream from a DTxA. However, the main transmitter is ***not*** a distributed slave transmitter, but rather is an ordinary ATSC transmitter. That means that the DTxP is transmitted intact over the air like any other transport data stream packet, complete with trellis/pre-coder states (i.e., they are *not* stripped out by the main transmitter). Upon reception at each translator site by a VSB receiver, the slave transmitter performs its usual synchronization and timing processing as if the DTxP data came directly from an STL rather than over the air. Just as in normal DTx systems, the slave transmitters at the translator sites strip out the trellis/pre-coder states and the DTxP Reed-Solomon correction bytes. Thus, each of the slave transmitters in the second tier transmits synchronized signals on the same RF channel, but different from that used by the main transmitter. What makes this situation straightforward is that there is no on-channel synchronization required for the main transmitter, only the single tier remote translators. That is, the DTxA only controls the second tier translators, and the main transmitter free runs, operating autonomously (i.e., acting like an STL). The only issue that requires a slightly different processing scheme is that the cadence sync signal (bit-inversion of the MPEG 47h sync byte) is missing as the RF signal, which has data segment sync replacing the MPEG sync byte, is received by the translator site VSB receivers and typically no cadence sync is output indicating where the data frame sync had been inserted. However, this slight problem is mitigated by the fact that the position of the frame sync is sent redundantly in the DTxP as a pointer byte indicating the number of *packets* since the last cadence (i.e. VSB frame) sync. Obviously, the DTxRs will use these pointer bytes when the cadence sync is found to be missing from the transport stream. The only difference in operation is that when a cadence sync is used in a DTx situation, it comes once per frame, and when a DTxP is used in a distributed translator situation, it comes less often as determined by the system operator (e.g., typically once per second). Therefore, this scenario is reasonably similar to a typical DTx System, except that an RF signal from the main transmitter acts as the STL. Of course, this distributed translator system, like normal translator systems, will only work if the received signal at the translator input is above threshold and experiencing error-free DTV reception.

This simple technique is limited to a "one hop" translator system. The slightly more complicated translator scenario occurs when a 2nd tier (or more) is added to the system. But if synchronization is required only ***within*** each tier (and *not* between tiers), then a series of DTxAs can be placed at the source end so that *each* one is adjusted to control one tier. While they will all use the same OMP structure to create their individual DTxPs, each DTxA (one for each tier) can have its own unique OMP packet *type* (8-bits are allocated for this in the DTxP structure, with up to 32 DTS translator tiers defined by ATSC). At each translator tier, the associated DTxPs are interpreted and the embedded trellis/pre-coder state codes are removed along with the Reed-Solomon bytes before transmission via the antenna. However, since a *special* DTV receiver is necessary to pass along the Data Field Sync data segment information from the last tier to the current tier for subsequent transmission, the slave transmitters in each tier *after* the first generally will use the Packet Number information in the DTxP address to their tiers to establish the Data Field Sync location in the MPEG-2 transport packet stream. Therefore, DTxTs can be independently controlled to be synchronized within each tier. Remember that when complete terrain isolation exists between

the various tiers of a translator system, there is no need to synchronize the system among the various tiers, just within each tier.

One last parameter to consider when discussing implementation of slave transmitter synchronization techniques is the insertion rate of the distributed transmission packet. There is no *minimum* DTxP insertion rate specified in the ATSC A/110B document (**Ref A1-4**). However, a minimum value can be considered that is a compromise between overhead data capacity and the recovery time for any transmitter that becomes unsynchronized due to a "glitch". For example, an insertion rate of one DTxP per second would use only 0.0078 percent of the overall data channel capacity. The *maximum* DTxP insertion rate is once every 312 data packets (i.e., once per VSB field), which is at a 41.3 Hz rate (24.2 msecs period). At this rate, 0.3% of the overall data channel capacity would be used. Since the DTxP carries data that is to be synchronized on a field rate, this synchronization data cannot show up at each transmitter more often than once per data field. The actual DTxP insertion rate is controlled by the station's service multiplexer since the DTxA only inserts synchronization packets when it sees the reserved OMP at its input with the appropriate PID value.

RF Watermarking Background

In a distributed transmission network, it is advantageous to be able to individually identify each on-channel synchronized signal present at a given receive site as well as any other conventional co-channel DTV signal *interferors*. This ability allows and facilitates DTx transmitter *in-service* amplitude level and timing delay adjustment and verification as well as provides a means to easily measure the propagation channel impulse response. However, this ability to distinguish each slave transmitter for both *identification* and *measurement* is not possible without turning them ON one at a time since each slave signal was intentionally made to be identical to all the others in the system so that they can be treated as quasi-static multipath echoes by DTV receiver equalizers.

One way to solve this identification problem is to assign specific, identifiable pseudo-random binary sequence (**PRBS**) codes to each *transmitter* and to the DTx *network* itself. These long, robust pseudo-random codes can be transmitted at a very low amplitude level "underneath" the VSB signal in the same RF channel due to their significant coding gain. They have negligible effect on DTV receivers, and provide diagnostic information to special receivers about the DTx transmitter (its *timing* and *amplitude* relative to the other transmitters) as well as the *channel estimation* of the signal propagation path from each transmitter. Additionally, interference identification and location finding can be accomplished with these techniques as well.

The unique network and transmitter identification codes are sent to each transmitter by the DTxA through the DTx network as part of the DTxP. Once received, these codes are used to create a *binary* symbol sequence that is modulated *synchronously* with the host 8-VSB symbols in such a way that ordinary DTV receivers are relatively unaffected by them. The "buried" transmitter and network identification signals are called RF "watermarks" (**WM**), they make use of spread-spectrum technology, which is why they are aptly called "buried spread spectrum" (**BSS**) signals (**Ref A1-4**). In this DTx context, spread-spectrum methodology is defined as very narrowband *baseband* data energy generated at the DTV carrier frequency that is distributed (i.e. spread) over a much larger frequency band. This is accomplished by use of long sequential noise-like pseudo-random codes that are subsequently buried underneath the desired DTV signal. At the special receiver, powerful *correlation* methods are used on the spread spectrum signal to retrieve the original narrowband specific codes.

In order to successfully use these RF watermarks, the binary codes must exhibit certain desirable attributes. For example, in order for them to be *uniquely* detected by special test equipment without causing interference to each other, these codes must be *orthogonal* to each other in order to have low *cross*-correlation values. This means that one transmitter code will be unlikely to be mistaken for another transmitter code. Another requirement for these *transmitter identifiers* (**TxID**) is that a large number of these orthogonal codes must exist from which to choose if a practical assignment is to be made by some regulating body to all transmitters on a region-wide or nation-wide basis. The other requirement is that they need to have large coding gain. This means that when an *auto*-correlation is performed, a very large value is obtained when the received code and stored code in the receiver line up, but when cross-correlation is performed with noise, other noise-like signals, or other codes present, the output is very low (i.e., they average close to zero). This is how these spread spectrum signals can reside far below the noise-like VSB signal and still be able to be decoded accurately.

The buried spread spectrum RF watermark signal can be created by using a 16-bit Kasami code sequence that is repeated almost 4 times (actually, 3.9 times since the 4th version is slightly truncated) throughout the ATSC data field, but ***NOT*** during the ATSC data frame or data field syncs. This sequence is created by *summing* three separate binary sequence generators, referred to as tiers, as shown in **Figure A1-7**. Each of the three generator groups (linear feedback shift registers) has a set pattern of "tap gains," and their outputs are summed together (modulo-2) before being modulated onto the RF carrier and buried beneath the VSB signal. These tap gains can be described by generator polynomials as follows:

Tier 1: $G_{(16)} = X^{16} + X^{12} + X^3 + X + 1$

Tier 2: $G_{(16)} = X^{16} + X^{12} + X^{11} + X^9 + X^8 + X^4 + X^3 + X^2 + 1$

$$\text{Tier 3:} \quad G_{(8)} = X^8 + X^7 + X^6 + X^3 + X^2 + X + 1$$

Out of the 40 total shift-register elements (i.e., 16 + 16 +8), 24 of them can be initialized with TxID data (Tier 2 and Tier 3). From **Table A1-3**, it is clear that, in the DTxP, each DTx network is identified by 12 bits (**Network Identifier Pattern**) and each DTx transmitter is identified by 12 bits (**Tx Address**). This provides over 16 million different sequences that are orthogonal to each other, making unique identification possible. **Table A1-4** shows the bitwise definitions of these shift registers.

Table A1-4 Kasami RF Watermark Code Sequence Generator Preloading Definitions

Shift Register	Tier 1	Tier 2	Tier 3
X^{16}	0	Tx bit 12	-----
X^{15}	0	Tx bit 11	-----
X^{14}	0	Tx bit 10	-----
X^{13}	0	Tx bit 9	-----
X^{12}	0	Tx bit 8	-----
X^{11}	0	Tx bit 7	-----
X^{10}	0	Tx bit 6	-----
X^{9}	0	Tx bit 5	-----
X^{8}	0	Tx bit 4	Net bit 8
X^{7}	0	Tx bit 3	Net bit 7
X^{6}	0	Tx bit 2	Net bit 6
X^{5}	0	Tx bit 1	Net bit 5
X^{4}	0	Net bit 12	Net bit 4
X^{3}	0	Net bit 11	Net bit 3
X^{2}	0	Net bit 10	Net bit 2
X^{1}	1	Net bit 9	Net bit 1

The RF watermark must be properly synchronized to the VSB signal. For instance, the shift register clock operates at the 10.762 MHz symbol clock rate of the host VSB signal and in phase with it so as to simplify watermark detection and to minimize VSB data eye closings. The PRBS sequences, which have no implicit framing structure themselves, are ***not*** transmitted during the VSB binary frame or field syncs and thus do not degrade VSB field synchronization at any relative RF watermark insertion level. The PRBS sequences are repeated approximately four times per VSB field, before being sent to the equivalent of a 2-VSB modulator that uses the same pilot carrier as the normal 8-VSB data. These sequences, just like the normal VSB data, are constrained by the same root-raised-cosine spectrum shaping filter. Therefore, the RF watermark signal fills the RF channel uniformly with a noise-like spectral characteristic and exhibits steep transition regions just like the 8-VSB signal. This requirement allows the test equipment that is to receive, process, and display the amplitude, timing, and channel impulse response to quickly lock up to the RF watermark signal by using the normal 8-VSB data field sync detection and clock synchronization methods. Since these two methods are very robust (operating down to 0 dB SNR), it is advantageous for the RF watermark test equipment to use the same methodology.

The RF watermark insertion *level* (or bury ratio) can be varied by changing its related variable in the DTxP (see **TxID Level** in **Table A1-3**) as shown in **Table A1-5**. There are 8 different level settings, including OFF. Bury ratio is defined as the ratio (in dB) between the total *average* power (in 6 MHz) of the primary 8-VSB signal and the total *average* power (in 6 MHz) of a buried spread spectrum signal (i.e., the RF watermark) that are sharing the same RF television channel. However, care must be taken when the largest RF watermark levels are used, not only for effects on the DTV receiver threshold degradation, but also for effects on the DTV transmitter's linear and non-linear feedback correction. The transmitter correction issue can be mitigated, however, by designing the exciter so that the known RF Watermark signal is accurately subtracted out from the incoming linear and non-linear feedback signals whenever the RF Watermark is active.

Table A1-5 Kasami RF Watermark Bury Ratio Definitions

Bit Value	Bury Ratio	Net Tx SNR w/30 dB Tx SNR	Rx Threshold
000	ID OFF	30.0 dB	15.04 dB
001	39 dB	29.5 dB	15.05 dB
010	36 dB	29.0 dB	15.07 dB
011	33 dB	28.2 dB	15.11 dB
100	30 dB	27.0 dB	15.18 dB
101	27 dB	25.2 dB	15.32 dB
110	24 dB	23.0 dB	15.63 dB
111	21 dB	20.5 dB	16.30 dB

Note: the 24-dB and 21-dB bury ratios should only be used in *off-line* testing

The RF watermark can also be modulated with slow speed data that is synchronous with the host 8-VSB data structure by inverting bit-by-bit the *entire* Kasami *sequence* to represent logic 1 and not inverting the entire *sequence* for logic 0. Since there are about 4 Kasami sequences per VSB data field, a total of 4 bits can be sent per VSB field, which is a 165.3 Hz bit rate (i.e., 4 x 41.327). If even more robustness is desired, "bits" (i.e., entire sequences) may be combined in pairs or in quads to gain 3 dB or 6 dB in exchange for ½ or ¼ the data rate, respectively.

Figure A1-8 illustrates the use of the RF watermark signal as it might be seen on a special piece of test equipment. Since DTxP information is transmitted over the air, with the exception of the trellis/pre-coder *states*, a piece of test equipment can find the DTxP, store the network and transmitters' RF watermark codes, and then search for any or all of the DTx transmitters. Note that the relative DTS *timing* and *amplitude* can be easily observed and measured from such a display. Therefore, special test equipment can individually display the signals received from various DTx transmitters and allow straightforward adjustment and optimization of DTxT timing and radiated power.

Since the Kasami codes have over 50 dB of coding gain, the fact that they are often placed 30 dB below the 8-VSB signal does *not* cause a problem since the coding gain allows them to be detected by special test equipment (via powerful correlation techniques) at least 20 dB *above* the 8-VSB signal energy. If time is allowed for averaging, the coding gain increases. Since each DTx transmitters have its own unique orthogonal spread spectrum Kasami codes, they have very little effect on each other's codes, which is the result of signal orthogonality. Each transmitter's signal response can be color-coded on a color display on a customized piece of test equipment to make it easier to interpret. Note that an advantage exists with this methodology in that these results can be obtained with all the transmitters on-line, that is, while the 8-VSB transmitters are still transmitting RF data to all the viewers in their service areas. The buried spread spectrum Kasami signals in these photos (see **Figure A1-8**) are about 27 dB below the VSB signal, so there is less than 0.25 dB threshold degradation in consumer DTV receivers.

Note that relative transmitter amplitude and timing are not the only useful information that can be obtained from this display. Since these curves are the result of correlation of the incoming RF watermark with the known respective Kasami codes for each transmitter, it can be seen that any propagation-induced multipath echoes of each transmitted signal will also correlate and show up on the plots. Therefore, the individual plots can also serve as propagation channel impulse responses for each transmitter. This is due to the fact that the RF watermark signals have the same spectral characteristics as the VSB signal and traverse the same RF propagation path as the VSB signal. Again, due to the orthogonality of the various Kasami codes, each correlated signal will show the impulse response from one transmitter, allowing the user to evaluate the amount of naturally occurring multipath in addition to the DTS-induced multipath from all the transmitters. Both multipath *amplitude* and *delay* can be easily measured. It is clear that even for *single* transmitters that are not part of a DTx network, transmitting an RF watermark signal can be very helpful to determining multipath conditions in critical reception areas, or, if a nearby station has the watermark as well, the amount of co-channel interference can be easily measured as well.

The effects of the RF watermark on the normal 8-VSB signal depend on the insertion level of the RF watermark. For most applications, a level of -30 dB can be used with little effect on the DTV receivers in the service area. **Figure A1-9** illustrates an idea 8-VSB data eye diagram that has a -27 dB RF watermark present. Note the extra 8 small eyes that exist because of the synchronous binary Kasami codes riding "underneath" the VSB signal. **Figure A1-10** illustrates the same condition, but shows the constellation diagram. Instead of 8 vertical lines, each of the 8 lines has been "split" into two lines again due to the synchronous *binary* (2-level) Kasami signal that is present. Nevertheless, the 8-VSB eyes are still fairly wide open, with little effect on the threshold of DTV receivers.

The RF watermarking technique appears to be a great method of making *on-line* field measurements of DTx relative timing delays and amplitudes as well as propagation conditions for both DTx and non-DTx situations.