CO-SITE INTERFERENCE MITIGATION FOR VHF COM VOICE AND DATALINK OPERATIONS

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Abstract

The use of datalink communication in the VHF avionics band is increasing. This expansion increases the opportunity for interference between a datalink transmitter and a co-sited conventional AM type receiver for voice reception. The interference is audible by the pilot using the AM receiver and takes the form of periodic or aperiodic noise bursts or clicks heard over incoming voice traffic. Signal processing techniques can subjectively eliminate or greatly reduce the interference caused by simultaneous voice and datalink operations.

This paper describes a method that has been used to demonstrate the technique.

Where We Are – The Current State

A typical air transport aircraft installation includes three VHF communication transceivers. Two of these transceivers are used for ATC and for AOC voice communication. The third transceiver is used for datalink operation. Until recently the datalink operation was for AOC use only. Recent regulatory and industry activities have proposed the use of VDL Mode 2 datalink for ATC in addition to AOC.

The voice transceivers conventional amplitude modulation. VHF datalink systems use various data signaling formats:

- ACARS uses amplitude modulation of a carrier with 2.4 kbps MSK modulation of a 1.8 kHz subcarrier. It is spectrally inefficient and has significant energy present outside the designated channel bandwidth of 25 kHz.
- VDL Mode 2 uses a differentially encoded 8-ary phase shift keyed signaling format (D8PSK) whose symbol rate is 10.5 ksym/sec. The spectral width of the main lobe is constrained to 16.8 kHz.
- VDL Mode 3 uses the same signal-in-space as VDL Mode 2, D8PSK., however it uses a TDMA channel access mechanism. The major frame rate is 120 msec and the burst length is approximately 20 msec. Although fully developed, VDL Mode 3 has not been deployed and can be considered deprecated at this time.
- VDL Mode 4 is a datalink system that uses 19.2 kbits/s GFSK. This TDMA system has 75 slots per second. VDL Mode 4 is required to meet the same adjacent channel spectrum requirements as VDL Mode 2 and 3. The burst duration for this system is around 13 msec. This system is undergoing trials and is not widely deployed.

The transmitted burst duration of these systems varies as they carry somewhat different data content. The ACARS burst can vary from 50 msec up to one second for a 200 character message. Due its transmission inefficiency, whenever possible ACARS messages are encapsulated and transmitted over the VDL Mode 2 system in a method referred to as ACARS over AVLC (AOA). This is probably the predominant use of VDL Mode 2 at this time – to carry encapsulated ACARS traffic. This may change if VDL Mode 2 is used for ATC traffic.

Some statistical data[1] of an airborne VDL Mode 2 transmitter with AOA traffic is shown in Figure 1. The histogram shows that the highest numbers of transmissions correspond to a burst duration from 10 msec to 60 msec. The time between transmissions was predominantly ten to thirty seconds.

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[1] A total of 40 logs from a FINAIR flight of VDL Mode 2, April 2009.

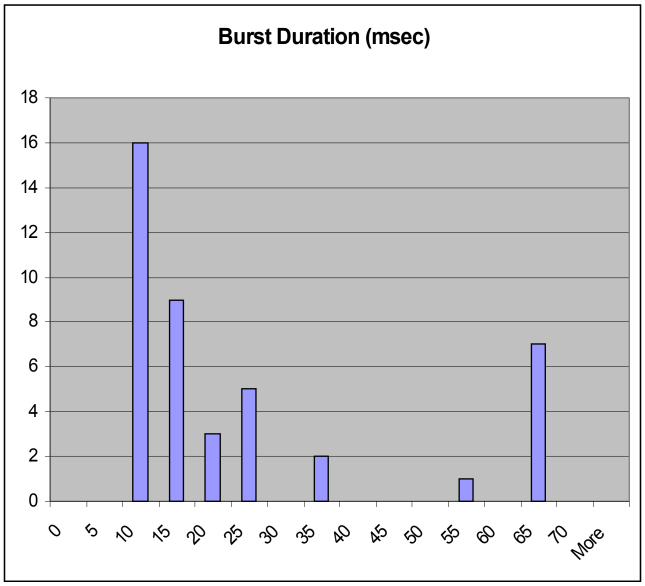

Figure 1. VDL Mode 2 burst duration histogram.

Co-site interference is always one of the design parameters of the airborne environment. Equipment specifications for large-signal performance normally correspond to interferers generated external to the aircraft. The performance in the presence of co-site transmitted interference is not critically specified due to the extremely high signal levels involved. The signal level of a co-site transmission measured at the RF input of a victim receiver can be as high as +5 dBm in Air Transport type aircraft. Regardless of the levels involved, interference immunity can not be specified when the interference is generated at the receive frequency.

The nature of the co-site interference depends on the frequency separation, antenna isolation, the characteristics of the interfering signal, and the design of the victim receiver. One source of audible interference is the transmitter noise floor associated with a co-site transmission. Even though the interferer's transmission frequency may be several MHz away from the receiver's tuned frequency, the noise power in the receiver's demodulation bandwidth[2] may be as high as -88 dBm at the antenna[3].

If the victim AM receiver is unmuted, as when listening to an incoming AM transmission, this sort of interference will be heard as an increase in background noise for a period corresponding to the duration of the transmission.

Another type of interference is caused by the dynamic effects of the high level interfering signal upon the operation of the victim receiver. Depending on the receiver's design, certain audio artifacts, clicks or pops, may be present due to effect of the nearly instantaneous change in the level of the interfering signal.

It is a subjective judgment as to whether an audible artifact is disruptive to communications, though periodic interference from VDL Mode 4 is much more noticeable than interference of a randomly occurring nature, as from VDL Mode 2.

Mitigation Technique

The method for reducing the effect of interference is implemented solely in the victim AM receiver. It takes advantage of signal processing techniques used in the audio recording industry for the restoration of damaged audio segments in an audio stream.

For this application, the audio present during the time of the interfering transmission is treated as damaged or missing audio. At a time corresponding to start of a scheduled burst, and for a duration corresponding to the length the burst, the victim AM receiver will replace the demodulated audio with an equal length sample of synthesized audio. There are several techniques discussed in the literature to synthesize or reconstruct the replacement audio [1] [2] [3]. The choice of a method is dependent upon its effectiveness and on the available digital signal processing power in the receiver. Two examples are: 1) replacing the audio with a sample of the audio that immediately preceded the replaced section, 2) replacing the audio with a sample of audio extrapolated by a statistical autoregression (AR) of audio both preceding and following the replaced section.

The first method is somewhat crude, even with cross fading between the replaced audio and its adjoining segments as there often exists an audible

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[2] The AM receiver's IF bandwidth is approximately 20 kHz.

[3] Based on 35 dB of co-site attenuation and -53 dBm maximum transmitted power beyond the fourth adjacent channel as specified in "Minimum Operational Performance Standards for Aircraft VDL Mode 2 Physical, Link, and Network Layer", RTCA/DO-281A, November 8, 2005. Section 2.2.1.3.7 Adjacent Channel Power.

phase shift at the boundaries of the segments. Tests have shown the limits of the technique to interferers that occur up to once per second and whose length is no more than perhaps 20 msec.

The second method is suitable for interferers that occur at higher rates and whose length is as much as 70 msec. While more computationally intensive, the second technique is much more effective at mitigating the co-site interference than the first.

Both methods require prior knowledge of the time of the start of a co-site transmission. Some time prior to the start of a transmission, the transmitter signals the victim receiver the starting time and duration of the transmission. In our test this information was conveyed via a data bus interconnecting the two co-sited transceivers. The victim receiver uses the transmission timing information to initiate the audio reconstruction.

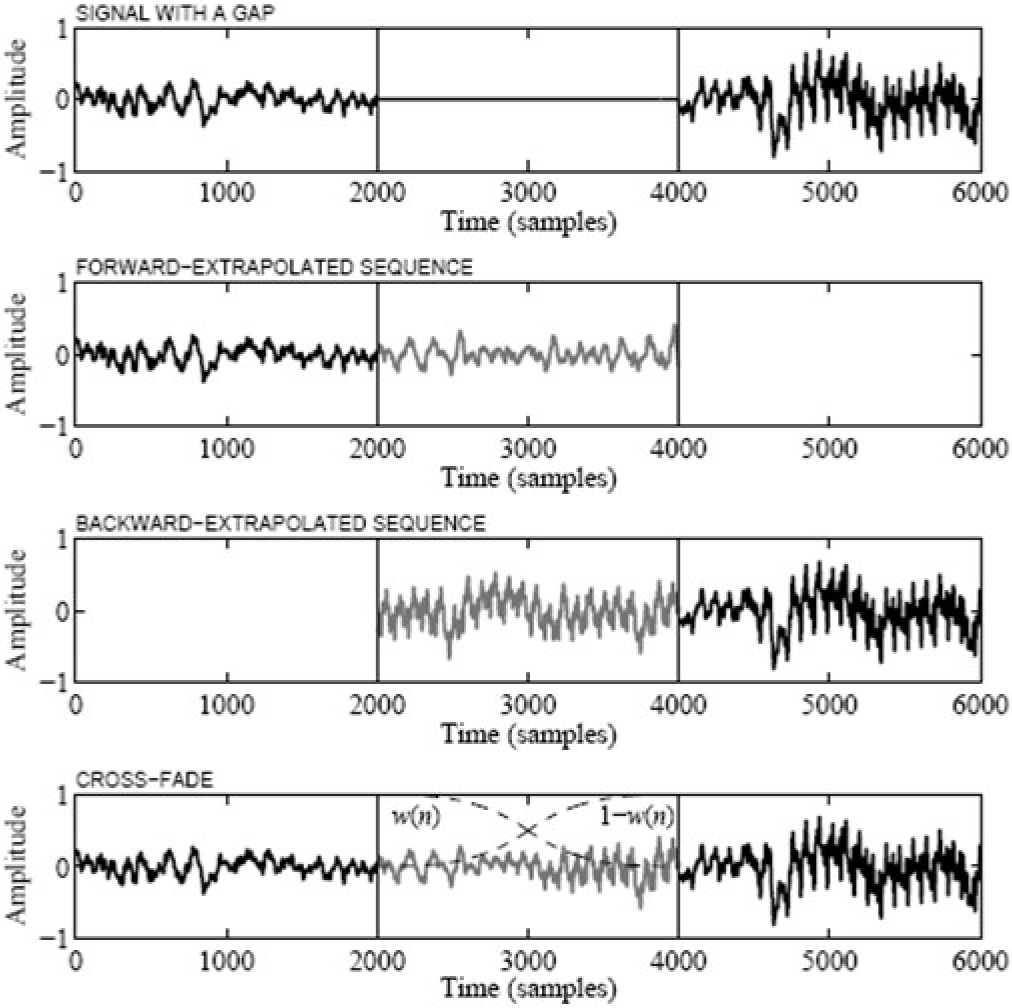

Figure 2. Digital audio restoration based on AR extrapolation.

Using the autoregression technique, the audio prior to the burst is analyzed to extrapolate a forward prediction of the audio. The audio following the burst is analyzed (reversed in time) to extrapolate a backward prediction of the audio. The two extrapolated samples are then mixed using a cross-fade function and the resulting audio is used to replace the noisy section. This is illustrated in Figure 2. Since the technique essentially replaces a section of audio, the audio stream can be thought of as having a gap or drop out into which reconstructed audio will be inserted. This is illustrated in the first row of the figure shown as a signal with a gap. The second row illustrates the first step in the generation of the replacement section; the sample of audio preceding the gap is analyzed and audio is extrapolated into the gap based on the characteristics of the preceding audio section. The third row illustrates the backward extrapolation performed using the analysis of audio following the gap. The last row illustrates the cross-fade function used to restore the audio section in the gap.

The time allocated for the preceding and following audio sample periods are variables of the process. The number of samples analyzed before and after the gap affects the quality of the reconstructed audio. It also affects processing delay, though certain signal processing techniques can be used to reduce the computational load. Our testing used 16 ksamp/sec audio and analyzed a sequence of samples before and after the gap equal to the gap length, up to a maximum of 70 msec.

Since the entire following block of samples must be stored before the analysis of that segment can begin, a static audio delay will be incurred before the reconstructed segment can be inserted. This delay will equal to the time needed to store the audio samples following the region to be reconstructed, analyze the following sample segment, (the analysis of the preceding segment can be begun as soon as it is stored), and to perform the cross-fade of the extrapolated audio. The processing delay is highly dependent upon software design, the order of the AR used for the analysis, and the available processing power. Based on the system design specification of the VDL Mode 3 digital voice system [4], the end-to-end delay should be less than 250 msec. Our demonstration software showed a processing delay of 180 msec with an AR order of 100.

Acknowledgements

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Biographical Sketches

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