Telex Communications, Inc.
Marketing Note 980301A

The Myth of the "All Digital" Intercom System
and Design Issues Related to Digital Intercom Systems

Introduction

There has been much discussion regarding the nature of an "all digital" intercom system or matrix, with a number of conflicting definitions and assumptions. This paper is intended to clearly define the issues, terms of discussion, and to set forth the Telex philosophy regarding digital matrices for intercom systems as implemented in the **RTS**™ **ADAM**™ and **ZEUS**™ Intercom Systems.

Human Factors

Human beings are analog creatures by nature - we speak words which cause amplitude variations in the air - that is to say we create sounds which are inherently analog. The same process occurs when we hear sounds, the varying sound pressure levels reach our ears as analog signals. For any digital processing of signals representing "sound", a conversion must take place external to the person making or listening to the sounds. This conversion is referred to as Analog to Digital (A/D) or Digital to Analog (D/A) depending on whether we refer to the encoding or decoding process.

Sound Quality

The quality of the sound when converted to or from the digital form is dependent on a number of factors including rate of conversion (sampling frequency), which determines bandwidth and resolution of the conversion (number of bits per sample). Broadly speaking, the higher the sampling frequency, the better the sound quality (44 kHz sampling is better than 22 kHz). The greater number of bits per sample, the better the sound quality (16 bit conversion is better than 14 bits).

Sound Processing

Once in the digital domain, a number of factors affect the preservation of sound quality. When mixing multiple signals (for listening to more than one person at a time), the digital mixer must preserve all bits - the addition of two 16 bits signals may yield a "result" with 17 bits - 32 signals (party-line perhaps) may result in a 21 bit result. Ideally the increased "data width" should be preserved until final conversion to analog to avoid digital noise and signal artifacts. Telex has designed the **ADAM**™ and **Zeus**™ Intercom Matrices with internal mixing bus width of 44 bits, and an external bus width of 24 bits to fully preserve all audio information through all parts of the signal selection, processing, and mixing stages.

<u>System Design Issues</u>

Now that we have terms of reference, we will discuss the practical implementation of these factors in a system design. As we have discussed in the **<u>Human Factors</u>** section of this paper, humans deal with sound as amplitude variations in air pressure. Microphones and speakers (or headset elements) convert these acoustic signals to and from ANALOG electrical signals. To utilize the benefits of digital signal processing the design question then becomes "where is the optimal system location to do the analog to digital conversion?".

Location of the required A/D and D/A circuitry - There are a number of possible locations in the signal chain for the converters - they can be located immediately before or after digital selection of a desired signal, or they may be located after all mixing and processing has occurred, but prior to transmission to external connections, or they may be located at the source and destination in an external chassis, or they may be located internally in the external device (keypanel or program source / destination).

The option of placing the converters immediately before or after the digital selection circuitry is not a viable solution as it would impose a requirement for analog processing and mixing - multiple converters and analog mixers to group signals would be needed - unacceptable from both a price and performance standpoint - the result would be an expensive, poorly performing system.

The option of placing the converters in an external chassis at source or destination is not particularly viable due to cost considerations - there would be extra chassis costs, extra power supply costs, interconnection costs, etc..

This leaves two options which are viable; the converters may be placed internal to the matrix whilst preserving "fully digital" selection, mixing, and processing, or the converters may be placed internal to the source / destination device, achieving a similar preservation of a "fully digital" matrix.

There are a number of price / performance / convenience issues which have varying importance in making a selection between the two remaining options.

The placement of converters in the source / destination device permits a single pair or coax to be used for interconnection between panel and matrix, and permits a claim of a system which is more "fully digital" than any of the other options due to the delayed conversion to and from analog. This option has the disadvantage of higher cost for the required mux / demux for audio and data, it has the additional disadvantage of requiring that all analog sources and all digital sources in other formats (AES/EBU/ SPDIF, etc.) be converted or handled by modified interfaces. This method additionally imposes a requirement for separate interface cards for inputs, outputs, and user stations - a user may find himself (or herself) with unused keypanel ports, while having a shortage of IFB connections, program inputs, or 4 wire connections - there is no ability to maximize the use of existing equipment - additional cards (and additional matrices!) may be required to meet operational needs.

This method has a theoretical advantage of reduced degradation of signal when connecting the external devices. In practice, the losses incurred in analog transmission of low impedance balanced signals between equipment are so low as to be considered negligible.

The placement of converters internal to the matrix, after full digital selection, processing and mixing has the advantage of permitting a single card to be used for interface to keypanels, analog audio sources, and destinations without requiring any external converters. It does impose a requirement for a three pair connection between matrix and keypanels, however, it has been noted that the vast majority of installations are “permanent” and the penalty for a one time installation of additional wiring imposed by a non-multiplexed analog scheme is usually minor compared to the advantages outlined above.

This is the methodology which Telex has settled on for the RTS™**ADAM**™ Intercom System. Advantages not mentioned above include a greatly reduced need for spares, total user flexibility of configuration of systems - i.e., at any time, any available port may be used for a keypanel, a 2w / 4w interface, a program in, an IFB out, or a monitoring output, all without the need for additional cards, converters, or rewiring.

SUMMARY

The factors of most influence in determining the audio performance of a matrix are the resolution and sampling rate for the A/D and D/A converters - not the placement of those converters. Telex has chosen to exceed existing systems performance by incorporating precision converters, operating at 44.1 kHz, with 16 bits of resolution in the **ADAM**™ Intercom system, - identical to the standards for CD audio; and unmatched 20 bit conversion at 44.1 kHz in the **ZEUS**™ DSP Intercom Matrix

The issues related to system installation and maintenance have been examined, and one time installation issues have been judged to be of lesser importance than ongoing reliability, flexibility, operator convenience, and “cost of ownership”.

Telex will continue to examine these issues as customer requirements and available technology change - always putting the customer benefits first.

“Exactly what you’d expect from a sound company like Telex”