System Levels Overview

EDACS® SYSTEMS

System Levels and Alignment Guide

ERICSSON

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NOTICE!

Repairs to this equipment should be made only by an authorized service technician or facility designated by the supplier. Any repairs, alterations or substitution of recommended parts made by the user to this equipment not approved by the manufacturer could void the user's authority to operate the equipment in addition to the manufacturer's warranty.

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1. INTRODUCTION

1.1 PURPOSE

The purpose of this manual is define the setup and align the audio levels in an EDACS Network. The manual

- provides an overview of EDACS Network Inter-Site Communications levels,
- defines the overall and survey requirements of a Link,
- describes the effects of Link Loss, and
- defines the alignment requirements of EDACS System levels.

In addition, this manual will define the more stringent requirements placed on levels for IMC Release 5 software.

NOTE

The intent of this document is NOT to define the alignment for a specific Network but for the user to understand the level requirements of their specific Network and make the necessary calculated alignments. This version of the guide will include more detail than is intended for future versions until that detail is incorporated into the relevant maintenance manuals.

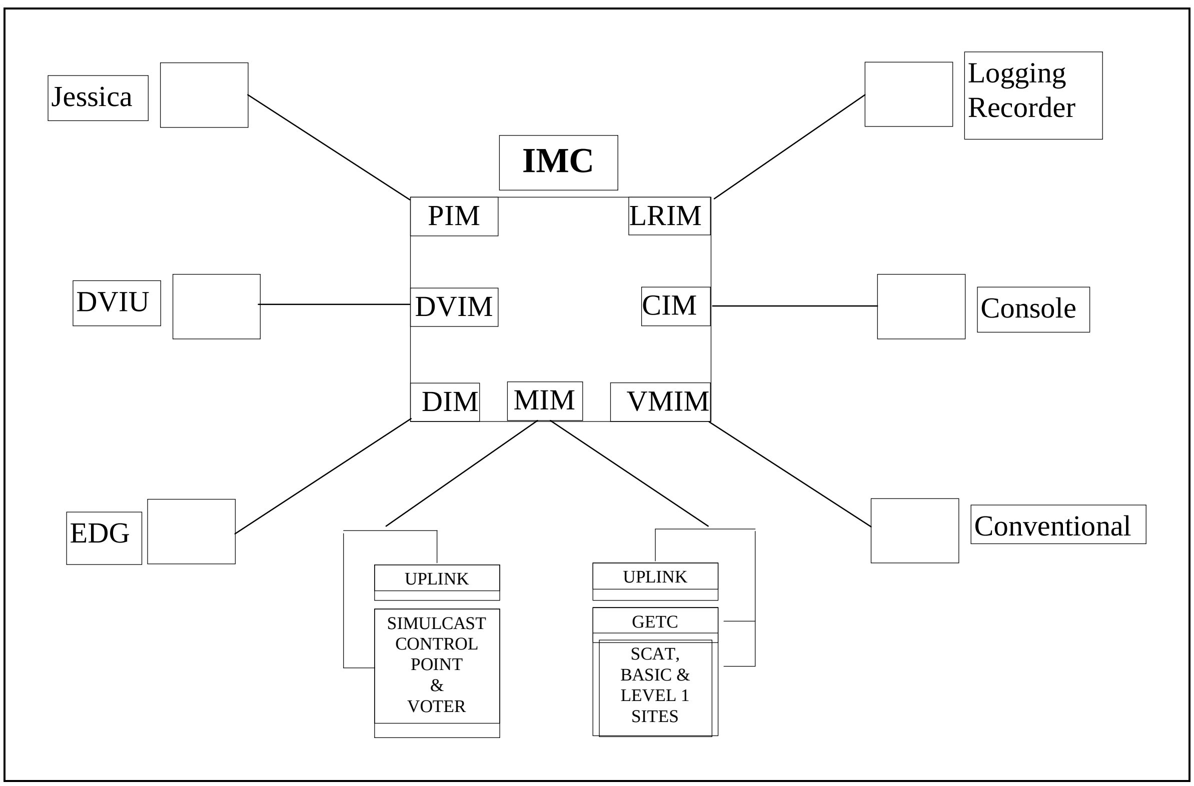
Any Proposal, Contract, Application or Upgrade System Design requires detailed examination of the links in an existing Inter-Site Communications System and/or the inclusion of a detailed Link specification to ensure that the performance of any link will not diminish the performance of the EDACS Network.

This document does not define the alignment for Terminal equipment. It assumes all terminals have been aligned to the same performance requirements defined for the type of Base Station system deviation, such as Wide Band, Narrow Band or NPSPAC.

1.2 SCOPE

This document applies to:

- Clear Voice Systems Designs,
- Digital Voice Systems Designs,
- Data-capable Systems Designs,
- Analog and T1/E1 link interface,
- Leased-Line, Microwave and Fiber-Optic link Designs.



This document addresses the following CEC/IMC interfaces:

- MIM Multi-Site Link - Site to/from IMC
- DVIM IMC to/from DVIU
- CIM IMC to/from Console
- LRIM IMC to/from Logging Recorder
- DIM IMC to/from EDG
- PIM IMC to/from Jessica
- VMIM IMC to/from Conventional System

1.3 RELATED MANUALS

As this document provides an overview to all components of an EDACS System, it relates to:

EDACS System and Site Installation Manuals:

- Conventional Station Installation Manual LBI-38636
- Basic and Level 1 Sites Installation Manuals AE/LZT 123 3242/1
- Simulcast Installation Manuals LBI-39206

EDACS Network Equipment Installation Manuals:

- CEC/IMC Installation Manual LBI-38662
- Console Installation Manual (depends on console type)
- DVIU Installation Manuals LBI-39041
- EDG Installation Manuals LBI-38962
- Jessica Installation Manuals LBI-39000

Vendor Manuals for any ancillary equipment incorporated in an EDACS System.

1.4 TECHNICAL SUPPORT

For Technical Assistance, contact the Ericsson Technical Assistance Center (TAC) at 1-800-592-7711 (804-592-7711 outside USA).

1.5 GLOSSARY

Refer to Appendix A ‘Glossary of Definitions’ for explanation of terminology and abbreviations used in this document. A complete listing of mobile radio terms and abbreviations is available by ordering ECR-1895.

2. SYSTEM LEVELS OVERVIEW

2.1 OVERVIEW

The link elements of an EDACS Network are shipped from the factory pre-configured and aligned for working input and output levels of 0 dBm Voice Peak (Vpk), (-10 dBm Average Voice). This assumes that the links in the network have no loss. If the links in the customer's system have loss, the equipment levels must be adjusted during installation and some equipment may require re-alignment.

Every piece of equipment in an EDACS Network has minimum and maximum level specifications for input and output. When inter-connecting the equipment, the ranges must be matched and the levels correctly aligned to ensure the source output level is sufficient to maintain a good Signal-to-Noise ratio and the dynamic range on the link is within the specifications of all the link components. If the signal is correctly aligned, both Multisite performance and Radio System RF performance will meet the System Design requirements.

This section will define the types of audio found on the links and the available ranges for each.

NOTE

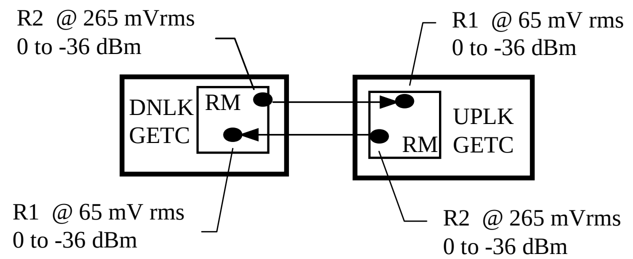
Rockwell Modem audio has different characteristics compared to Clear Voice. Modem audio has peaks that, when measured with an average responding RMS Meter (TIMMS), are about 5 dB higher than the level indicated on the meter. To compensate for this, adjust the Rockwell modem output to a peak level 5 dB below that required for Clear Voice.

2.2 EDACS AUDIO

There are three types of audio which must be routed between EDACS Radio System, IMC, and Network for Multi-Sited Systems:

- **Downlink Modem Audio**
  Data Modem audio between Downlink and Uplink GETC
- **Base Station Clear Voice Audio**
  Voice audio between Base Station and either another site Base Station and/or a console via the IMC
- **Base Station GETC Modem Audio**
  Modem Audio between Base Station GETC and either another site Base Station GETC and/or a DVIU and/or an EDG via the IMC.

2.2.1 Downlink Modem Audio



Uplink-Downlink Modem level alignment

The Downlink Modem Audio has a unique path from the other communications. The Downlink and Uplink GETCs are aligned for a nominal output level, in a loss-less link, 15 dB below the Vpk level specified for the link, and for a nominal input adjusted to -22 dBm; i.e., for a Link Vpk level of 0 dBm, adjust the Modem level to -15 dBm. Note that this input level is measured on the output of the op-amp interfacing the input line to the modem input. Once powered on, the modems train and remain synchronized.

NOTE

Rockwell Modems, having a wide acceptance range of input and output levels, are reasonably resilient to link-loss. Take care with the peak value of the level to ensure that the audio is not clipped in the link and the receiving modem will operate correctly with a wide range of loss.

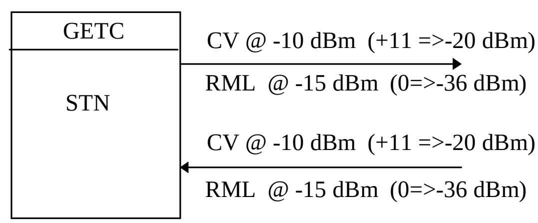
Each GETC converts digital voice or data into modem audio, transmits the data to the other GETC which reconverts the modem audio to digital voice or data. If the multisite link connecting the Site to the IMC has loss, these levels may be adjusted over a wide range to compensate for link-loss. This range is 0 to -36 dBm.

The modem audio output of the source GETC is aligned with (R2), within the specifications of the link, to a level that achieves a good Signal-to-Noise ratio on the link and that is within the acceptable dynamic range of the destination GETC modem. The destination GETC modem can provide gain or attenuation (R1), as necessary, to the incoming audio to provide the correct input level.

2.2.2 Clear Voice Audio

For Clear Voice systems, Base Station and IMC communicate via a 4-wire link with nominal input and output levels of -10 dBm. Base Station equipment is factory aligned to provide 60% System Deviation with an Average Voice Test Tone (AVTT) input level of -10 dBm and an AVTT output level of -10 dBm for a received RF signal.

Base Station Clear Voice Audio terminates at the IMC where the source audio is digitized, routed on the TDM Bus to the destination interface, converted to analog audio and amplified to the input requirements of the second link.



Base Station and GETC Input and Output levels
Using nominal -10 dBm AVTT requirement and -15 dB RML

The Base Station Clear Voice Audio Output is aligned at the Base Station for a level having good Signal-to-Noise ratio at the IMC. The good quality audio must be sufficient to overcome link loss and be within the dynamic range of the link equipment to prevent audio clipping.

The Base Station Clear Voice Audio Input is aligned at the IMC to provide an audio input to the Base Station sufficient to drive the Station to 60% System Deviation. The audio level must be sufficient to overcome link loss and be within the acceptance range of the link equipment to prevent clipping.

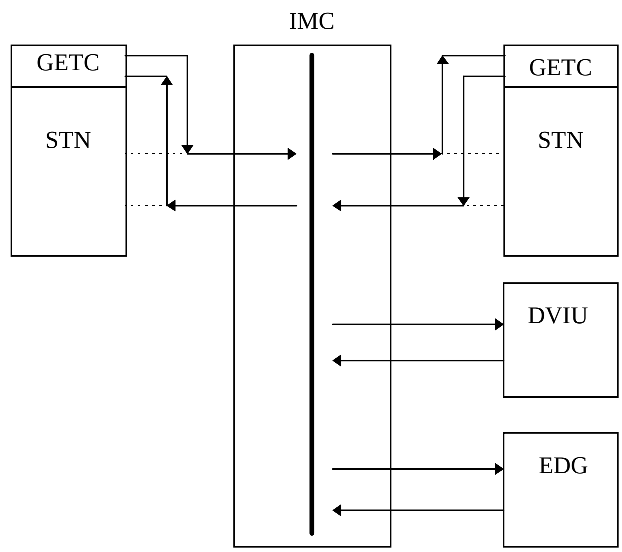
For systems using tone control, the **Securit** tone is at the level of Voice Peak (Vpk), **Function** tone is at the level of AVTT (Vpk - 10 dB) and the **Hold** tone is 20 db below AVTT (Vpk - 30 dB). This necessitates a 30 dB dynamic range requirement to be considered when defining the link and link equipment.

For systems using DC Control, the link equipment must be able to support Type 1 E&M Signaling. See MASTR III Base Station LBI-38636 and System Change Document for CEC/IMC T1/E1 Interface LBI-39107 and 39108 for detailed information.

Both Base Station and IMC equipment have a wide range of acceptable input and output levels which can be adjusted to compensate for Link Loss. If the multisite link connecting the Base Station to the IMC has loss, these levels must be carefully adjusted to ensure the IMC provides the Base Station with sufficient audio level to provide the desired system deviation, and the Base Station provides sufficient audio level to meet the IMC's requirements.

2.2.3 Base Station GETC Modem Audio

Base Station GETC Modem audio carrying digital voice or data communication is routed from the Base Station to the destination equipment via the IMC Network. At the multisite link, the modem audio uses the same physical path as the Clear Voice audio. Destination routing depends on whether the call is Clear Voice, Digital Voice or Data.



Modem communication across the IMC Network

The Base Station GETC Modem Audio Output levels are normally adjusted for a measured level that is 5 dB below Clear Voice. Note the -10 dBm Modem audio has higher peaks than -10 dBm of Clear Voice audio. However, the IMC provides the same gain to both Modem Audio and Clear Voice Audio and Modem Audio will be output at the same level as defined for the Clear Voice on the second link.

If Modem audio in the system is to be set to a level below that of Clear Voice, the DIFFERENCE between Clear Voice and Modem audio must be defined for the poorest link (most loss) in the system. Additionally, the difference between Clear and Modem Audio levels must be the same for ALL Multisite links in the system. In this way, modem levels from various multisite links reaching the destination equipment will be at the same levels with respect to Clear Voice.

The modems have a wide input and output acceptance range but it is possible that, should the link be switched to a new route, especially in the case of Leased-Line links, more loss may be introduced that will put the signal at a level outside of the acceptance range of the destination equipment. As the DVIU and EDG must communicate to each of the Base Stations at each of the sites connected to the IMC, and as those links may vary in levels of loss, the Multi-Site Modem Data levels must be aligned, as for Clear Voice, to the performance of the link with the most loss. If not, the link with most loss will not be reliable for Digital and Data communications. Multi-Site Modem Data equipment should therefore be carefully aligned such that extra loss does not prevent successful communications.

If the link loss is too great, it is possible that the modem audio will be at too low a level to maintain an acceptable Signal-to-Noise ratio on the link. The modems may not maintain synchronization, modem audio may be lost, there may be a difference in performance between multisite calls perceived at either DVIU or EDG, or that Multisite and Network performance will vary from site-to-site. If this is the case, either the link must be changed or the audio must be amplified at the input to the link.

The Base Station GETC Modem output should be aligned for an average measured level that is 15 dB below the specified Vpk allowed for the link and the input alignment should attenuate the input level to -22 dBm. For a Link with a specified Vpk of 0 dBm, adjust the Base Station GETC Modem output to -15 dBm. If Clear voice is set to a output level of -10 dBm, the GETC Modem audio output level should be set to -15 dBm.

3. INTER-SITE COMMUNICATIONS LINKS

3.1 INTRODUCTION

A "link" is defined as the path between two devices, each at different sites/locations, which will provide carrier service for EDACS System Audio and Data Inter-Site Communications. A link may comprise various stages of separate links, i.e. the path may comprise a microwave hop from one site connected into a leased-line stage followed by a hard-wire direct connection terminating at the second site. The requirements and specifications given for the link must treat the link as a whole and encompass all stages of that link.

This section will examine Inter-Site Communications Links, the requirement for examination of the link system and how to apply the results of survey:

- Link Requirements
- Link Types
- Link Characteristics
- Link Survey
- Output for Alignment

3.2 LINK REQUIREMENTS

The overall requirement of a link interconnecting the audio paths between two sites or locations is to ensure that the EDACS System meets contracted performance criteria. To meet this requirement, the link must:

- accept the full range of audio between the two sites,
- provide little or no loss across the link,
- provide little or no noise to the audio,
- provide little or no distortion to the audio.

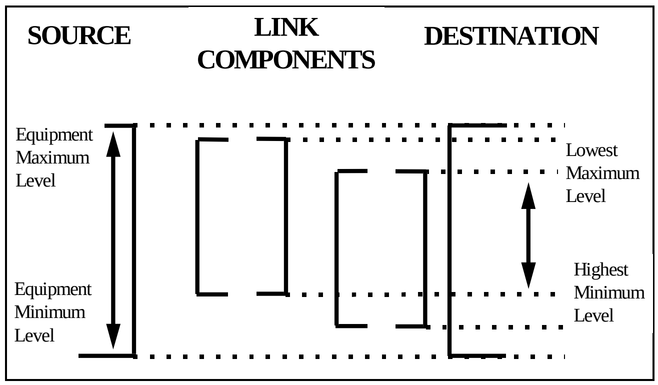
3.2.1 Audio Acceptance Range

Each stage of the link, including the equipment connected to the link, has maximum and minimum acceptable input and output audio level range specifications. The definition of EDACS audio for a particular link requirement is given earlier in this document and is shown in detail in Appendix A and B. The links must be examined to ensure that they will cater not only to the levels for EDACS audio but also to the signaling tones used for EDACS communications:

- **Securit** tone at 10 dB above AVTT (at Vpk)
- **Function** tone at the level of AVTT (Vpk - 10 dB)
- **Hold** tone at 20 db below AVTT (Vpk- 30 dB)

This entails a requirement for a 30 dB dynamic range over and above the specification for the AVTT level to be used for the link and considered when defining the link and link equipment.

The diagram below is a graphical representation of how to determine the acceptance range of a link comprising a source equipment, two link components and a destination equipment. Each step in the link has it's own acceptance range and, when considered in conjunction with all steps, the range of the whole is the calculation of lowest maximum and highest minimum in the link.



3.2.2 Loss

Each stage of the link, including the equipment connected to the link, has a defined level of link loss expressed in terms of dB Gain. The gain, or signal attenuation, for the entire link is the sum of the gains for each stage; i.e., if the gain for stage 1 is -3 dB and the gain for stage 2 is -16 dB, then the maximum gain for the link is -19 dB or a total signal attenuation of 19 dB.

The specifications for each stage of the link must be examined and/or tested to evaluate the level of loss provided. Tests are defined later in this section.

Some Link loss issues may be compensated for by the application of in-line audio amplifiers. Ensure that Leased-Line links do not exceed the maximum loss specified by the carrier - If at all possible, it is preferable that Bell 43202 Type 5 Data Grade Line Specification to the Phone Company is modified with the proviso that the Line loss must not exceed 10 dB.

3.2.3 Noise

Each stage of the link has a level of link noise expressed in terms of dBmC. The level of noise for each component in the link and for the link as a whole must be evaluated and confirmed to be acceptable for EDACS operation. Link noise is comprised of three types of noise;

- **Notched noise:**

  the noise present in a channel in the presence of a test tone.

- **Idle noise:**

  the noise present in a channel without signal audio.

- **Impulse noise:**

  a determination of periodic noise 'spikes' in a channel, in the presence of a test tone, for a defined duration.

3.2.4 Link Distortion

In any link, there may be loading of the signal at specific frequencies, normally caused by loading coils and associated circuitry attenuating certain frequencies in the audio. Such distortion may affect operation of modems and if dramatic attenuation is observed, the link provider is to be advised that the link is unacceptable.

To test for Link Distortion, a Bridge Tap and Loading Coil Detection test is performed in which the transmit audio into the link is slowly swept across the frequency range and the receive audio is observed to detect dramatic variations in response.

3.3 LINK TYPES

T1 or E1 via either Microwave, Fiber-optic cable or Leased T1/E1 generally have little or no loss and in these cases, alignment is straight forward. The multiplexers used normally have switchable input/output ranges and link attenuation settings, but the specifications of even these paths must be examined to ensure that they provide the level of service required.

If, however, the link is either Leased-Line provided by a Telephone Carrier, conforming to FCC 43202 (old specification - Bell 3002 Data grade) or equivalent, or is of any type but has a high level of loss, the specifications and characteristics of the link must be examined to ensure that the link loss will not adversely affect the EDACS System.

3.3.1 Characteristics

The minimum specifications for a Leased-Line link are as specified below. If more stages are connected to the link, these specifications apply to the entire link.

Frequency Response	1000 Hz	Reference
	500-2400 Hz	-1 to +3 dB
	300-2700 Hz	-2 to +6 dB
Long term Variation		4 dB
Maximum Noise		31 dBrnC
Maximum Frequency Error		±5 Hz
Maximum Net Loss		16 dB
Maximum Group Delay (800-2400 Hz)		2000μS
Minimum S/N Ratio		24 dB
Voice Audio levels Vpeak		+10 to -20 dBm
Modem Audio levels Vpeak		0 to -36 dBm
Tone Control (1050 - 2050 Hz)		0 to -30 dBm
Termination Impedance		600 Ω

3.3.2 T1/E1 Digital Data Rates and Hierarchies:

Digital signals have long been standardized in their electrical formats, bit rates and frame structures. The North American standard (ANSI) and European Standards (CEPT) were developed independently and coexist internationally (ITU/CCITT) as the major digital signal standards. Recently different hierarchies have been standardized for asynchronous networks, and these must be distinguished from the existing hierarchy for asynchronous network.

Appendix D includes tables providing a list of the standard asynchronous and synchronous digital signal hierarchies including the specifications for T1(DS1) and E1.

3.4 LINK SURVEY

There are many different reasons why Link surveys may be required, i.e.

- the Inter-Site Communications System already exists prior to installation of the EDACS System,
- the specifications for the customer's Inter-Site Communications System are unknown,
- Clear or Digital Voice alignment requirements for a link cannot be met,
- audio performance is unsatisfactory from a site,
- any part of the system is being upgraded and the audio specifications for the new equipment differ from that of the old equipment.

If no Inter-Site Communications System yet exists, i.e., at Proposal Stage, the engineer can issue the Link specifications shown in this document.

Existing Inter-Site Communications Systems should be surveyed during the Proposal Phase where possible and Link Requirements should be both included in the Proposal and confirmed with the customer during Contract System Design

3.4.1 Survey Preparation

The engineer investigating the Inter-Site Communications System must identify and understand the communications requirement of each link. He must define the specifications of the Link in terms of loss, noise, etc. and will therefore be able to calculate the input and output levels required of both source and destination equipment. Finally, he must provide the calculated levels and alignment information to the installation personnel.

The communications requirement of each link have been explained in this document. The engineer must review the system design to understand the features of the system, i.e., whether the system is Analog or Digital voice, whether Dispatch, Logging Recorder and/or Interconnect requirements include digital capability, and whether the system requires Data capability.

With this information, the engineer will be able to review what type of communications is required on which link.

3.4.2 Link Tests

The specifications of a Link to be used in an EDACS System have been explained in this document. If the Inter-Site Communications System already exists, the engineer must review the link specifications of that system to ascertain whether the system meets the performance criteria of an EDACS System. With this information, the engineer will be able to advise the customer whether changes or modifications are required. See Appendix C for full link test procedure (or get from TechMemo library). The procedure for measuring link loss and dynamic range is provided below.

3.4.2.1 INPUT AND OUTPUT ACCEPTANCE RANGE

The engineer is to identify the component parts of the Link from site/equipment interface (normally a punchblock) and is to define the Maximum and Minimum Input and Output levels allowed for each. He is to evaluate the lowest maximum and the highest minimum levels acceptable through the link to define the Acceptance Range. Level measurements are in dBm using a test tone to simulate average voice (AVTT).

If specifications are not available, the engineer may use the following procedure to evaluate the acceptance range:

1. Using a Audio Tone Generator, terminated in 600Ω impedance, generate a tone of 1 kHz at a level of -10 dBm. Connect to a Audio Tone Test Set, terminated in 600Ω impedance, and calibrate the Audio Tone Test Set for the generated value.

2. Isolate the link from the equipment at either site on the link, connect the Tone Generator to one end of the link and the Audio Tone Test Set to the other. If the capability is available, loop the link back at the destination end of the link and perform all tests in the one site.

3. Increase the transmit level until the received signal becomes distorted. Note this maximum level. Return the level to the calibration level above.

4. Decrease the transmit level until the signal again becomes distorted or too noisy to be workable. Note this minimum level. Return the level to the calibration level.

The two measured levels, although not exact, will give an indication of the acceptance range of the link.

The engineer is to document the specified range in terms of both Voice Peak and Average Voice for use by the Installer/Verification Engineer.

3.4.2.2 LINK LOSS

The engineer will identify the Link interface, will inject tones with known levels, will test the level of each tone at the output of the link and will evaluate the performance of the link. If it is possible, a loop should be placed at one end of the link to allow both tone injection and measuring from a single site making evaluation easier.

1. Using a Audio Tone Generator, terminated in 600Ω impedance, generate tones at 300 Hz, 1 kHz and 3 kHz, each at a level of -10 dBm. Connect to a Audio Tone Test Set, terminated in 600Ω impedance, and calibrate the Audio Tone Test Set for the generated value.

2. Isolate the link from the equipment at either site on the link, connect the Tone Generator to one end of the link and the Audio Tone Test Set to the other. If the capability is available, loop the link back at the destination end of the link and perform all tests in the one site.

3. Confirm that the received tone levels are the same as those transmitted. If they differ, calculate the how much loss is in the link path.

The engineer is to document the findings for use by the Installer/Verification Engineer.

3.4.2.3 NOISE

Each of the three types of noise is to be tested for the link.

3.4.2.3.1 Notched Noise

Notched Noise is tested, in both directions, by sending a test tone from Site A to Site B and measuring the noise and then reversing the direction of test to measure from Site B to Site A. Measurements are in positive units of dB relative to -90 dBm and are C weighted.

Follow the tests defined in Appendix C.

3.4.2.3.2 Idle Noise

Idle Noise is tested, in both directions, by measuring the level of channel noise from Site A to Site B and then reversing the direction of test to measure from Site B to Site A. Measurements are in positive units of dB relative to -90 dBm and are C weighted.

Follow the tests defined in Appendix C.

3.4.2.3.3 Impulse Noise

Impulse Noise is tested in both directions by measuring any impulse noise from Site A to Site B in a 15 minute period and then reversing the direction of test to measure from Site B to Site A for the same period.

The defined duration is 15 minutes and tests must be performed for -21 dBm0, -24 dBm0 and -27 dBm0 thresholds. Measurements are in counts of impulse instances in a 15 minute period and must not exceed 15 counts for unconditioned lines or 5 counts for D-type conditioned lines.

Follow the tests defined in Appendix C.

3.4.2.4 DISTORTION

Distortion is tested in both directions by performing a slow sweep in the generated frequency, from 404 to 3404 Hz, and observing the received signals for any sudden change in received level. The frequency response across the entire specified range is expected to be fairly flat.

If drop-outs are observed, the frequencies at which they occur are to be recorded together with the level of notch.

Follow the tests defined in Appendix C.

3.4.3 Survey Results

Following the survey of all the links in the EDACS System, document the test results for each of the links and calculate the EDACS equipment input and output signal levels for use during either installation or realignment.

Ensure that any link that has loss in excess of 16 dB, excessive noise or unacceptable distortion is carefully rechecked, that the customer/provider is notified that the link is unacceptable and must be either redesigned or realigned.

Store this Survey Results document in the project notebook and provide a copy of this information to the Installer or Technician who is to perform the realignment.

3.5 SURVEY OUTPUT

With the information from the survey, define the input and output Clear Voice and Modem levels for each link in the system. Use the equipment levels drawing in this document to ensure that the levels are within the maximums and minimums shown.

This includes:

- Clear Voice Line Input Level to the Base Station from the Multisite Link.
- GETC Modem Line Input Level to the Base Station GETC from the Multisite Link.
- Base Station Line Output level to the Multisite Link.
- GETC Modem Line Output level to the Multisite Link.
- Calculation of Input and Output ranges of the Multisite Link equipment.

If the survey is for additional links to be added to an existing EDACS System and the existing links have the GETC modem levels aligned at a difference from Clear Voice, review the effect of that difference for the newly surveyed link with most loss. If the new link will prevent a similar alignment, the system will need to be realigned to bring the Modem levels closer to that of Clear Voice.

4. ALIGNMENT REQUIREMENTS

4.1 GENERAL

EDACS Networks have varying Application design requirements; i.e., Clear Voice only, dual Clear and Digital Voice, Digital Voice only, Encrypted Voice, Data only or Voice-with-Data. As there are different setup requirements for each type of system design, there is a need to fully understand the alignment requirements of each type of communications.

For the purpose of defining alignment scenarios, this document defines three applications of EDACS System:

- **Clear Voice**,
  Only Clear Voice
- **Digital Voice**,
  Dual Clear and Digital Voice, only Digital Voice and/or Encrypted Voice
- **Data**
  Only Data or Voice-with-Data

This section examines the alignment requirements of the components parts of the three types of EDACS Networks.

The introduction of IMC Release 5, with it's inherent reduction of output audio dynamic range for Interface Modules (reduced from +13 => -25 dBm to +13 => -16 dBm), together with the use of T1/E1 interface cards, requires more care in the specification and examination of new or existing Inter-Site Communications links and in system alignment.

Any Proposal, Contract, Application or Upgrade System Design requires detailed examination of the links in an existing Inter-Site Communications System and/or the inclusion of a detailed Link specification to ensure that the performance of any link will not diminish the performance of the EDACS Network.

NOTE

All levels referred to in this document are measured at Average Voice Test Tone (AVTT) dBm level unless otherwise stated.

AVTT is 10 dB below Voice Peak.
RML is 15 dB below Voice Peak

4.2 CLEAR VOICE SYSTEMS

A Clear Voice System is defined here as a System without Digital or Data applications. This section defines the requirements of a multisited system with Dispatch, Logging Recorders and Interconnect.

4.2.1 Multisite Clear Voice Calls

A multisited, Trunked Radio-to-Radio Clear Voice call is between two or more EDACS Trunked Radio systems (sites) across the IMC Network.

The Base Station provides Line Output Clear Voice audio to the IMC (MIM) interface which furnishes the appropriate gain to the signal, provides A-D conversion and places the digital information on the TDM Bus. The IMC routes the digital information to the appropriate site interface, provides D-A conversion and amplifies the audio as required prior to routing the audio to the appropriate Site. The destination site receives the audio on the Station Line at the defined level for the link, modulates the transmitter and achieves the required system deviation. The return path is the same whereby the IMC directs the call to the appropriate Site Base Station.

If the destination site is conventional, the path is much the same with some path differences for CNI and CI.

The Multi-Site link alignment requirements for Clear Voice Systems are simple. The Base Station Audio Output should be aligned to provide the necessary IMC Input and the IMC will provide the necessary gain to the input prior to providing A-D conversion of the signal and placing it on the TDM Bus. The IMC Output should be aligned to provide the necessary Base Station Audio Input and the Base Station will be aligned to provide 60% of System Deviation for that given input.

The Base Station Clear Voice audio output should be aligned to compensate for link loss and provide an input to the IMC at a nominal level of -10 dBm. The Base Station can accommodate an output range of +11 to -20 dBm and the IMC can accommodate an input range of +12 to -25 dBm. The IMC Clear Voice audio output should be aligned to compensate for link loss and provide an input to the Base Station at a nominal level of -10 dBm. The Base Station can accommodate an input range of +11 to -20 dBm and the IMC can accommodate an output range of +13 to -16 dBm.

For a Clear Voice system, the Station GETC Modem output levels should be set at a level no more than 5 dB below that of Clear Voice.

4.2.2 Dispatch

Dispatch is defined as a Console operator communicating with one or more EDACS Trunked Radio systems (sites) across the IMC Network.

The IMC routes the digitized audio on the TDM Bus to the Console interface (CIM), provides D-A conversion and routes the Clear Voice audio, at the level defined by the Console Volume Bar, to the Dispatch equipment. The Console volume bar, which controls the CIM output level, is set to mid-level as seen on the Console display and the EAE is then aligned for that incoming level. The CIM has an output range of +13 to -16 dBm. The Console output is aligned for a level that is acceptable to the input range of the CIM, +12 to -25 dBm.

Console Transmit audio is routed to the IMC at a level calculated for the link and the Voice Peak equivalent of that level is programmed into the IMC Manager. The IMC provides the necessary gain to the signal, provides A-D conversion and places the audio on the TDM Bus for routing to the destination equipment.

4.2.3 Logging Recorder

Logging Recorder function is defined as a Logging Recorder receiving audio from groups or individuals across the IMC Network.

The alignment requirements for analog Logging Recorder operation are simplified in that they closely follow those of Analog Dispatch. The IMC routes the digitized audio to the LRIM as specified in the IMC Manager's Logging Recorder configuration screen, provides D-A conversion and provides the audio to the Logging Recorder at a level defined in the LRIM configuration screen.

The LRIM output level range is +10 => -16 dBm and this level must be aligned to that required of the Logging Recorder specifications.

4.2.4 Centralized Interconnect

Centralized Interconnect is defined as either Console or Radio users communicating with the Jessica Interconnect equipment across the IMC Network. Refer to LBI-39000.

The Jessica Interconnect system currently connects to the IMC with an analog connection. (A direct T1/E1 connection between IMC and MD110 is being developed and will utilize the TEC Interface.)

The IMC routes digitized audio on the TDM Bus to the Jessica Interface (PIM), provides D-A conversion and then provides gain to the audio as defined in the IMC Manager's PIM Configuration screen. The PIM routes the analog audio to Jessica multiplexer which provides A-D conversion prior to routing the audio to the Jessica MD110 PBX.

The IMC Manager has the following Voice Peak input and output ranges:

input:	+12 => -25 dBm
output:	+13 => -16 dBm Voice Peak

The Jessica equipment has different multiplexers, and therefore different input and output ranges, for T1 and E1 applications. The T1 multiplexer is NEC ND4 and the E1 multiplexer is ANT-Bosch. The ranges are as follows:

NEC ND4:	input = 0 dBm	output = 0 dBm.
ANT-Bosch:	input = 0 dBm	output = -14 dBm.

The alignment requirements are simplified in that the Jessica equipment is normally co-located with the IMC. If Jessica is remotely located from the IMC, pay attention to the level setup to conform to the requirements of the particular multiplexer.

4.3 DIGITAL VOICE SYSTEMS

A Digital Voice System is defined here as either a Clear/Digital Voice mixed System, a Clear/Digital Voice system using encryption or may be wholly Digital Voice. The difference between Clear Voice and Digital Voice is the additional A-D conversion and Encryption/Decryption steps provided by the DVIUs for Clear Voice destination equipment. The destination equipment includes Trunked Base Station, Conventional Channel, Dispatch Consoles, Logging Recorders or Jessica (CTIS).

In a Digital Voice System, the Base Station receives and demodulates the digital call, routes the data via the 'Volume Squelch High' line to it's GETC and the GETC converts the data to modem audio. The Control Channel information about the call sets up the multisite requirement linking the Base Station to the appropriate destination equipment via the IMC, both source and destination equipment modems synchronize and the data is transferred. The return path is similar as the Base Station GETC modem receives modem audio, converts the audio to data and transfers that data to the Base Station transmitter as High Speed Data. The transmitter is modulated and the destination radio receives the call.

4.3.1 Multisited Trunked Radio-To-Radio Digital Calls

A multisited, Trunked Radio-to-Radio Digital call is between two or more EDACS Trunked Radio systems (sites) across the IMC Network. The destination Base Station

GETC modem will synchronize to the Base Station GETC modem at the source site via the IMC and will receive the data. The IMC MIM furnishes the same level of gain to GETC modem audio as it does to Clear Voice. The IMC MIM provides A-D conversion of all audio and places the digitized audio on the TDM Bus. The IMC routes the digitized audio to the appropriate destination site MIM(s) which provides D-A conversion, amplifies the audio as specified for the secondary multisite link output level requirement and transmits the modem audio to the destination Base Station GETC modem. The return path is the same.

Note that the difference between GETC Modem and Clear Voice audio must be maintained across all the multisite links. This will ensure that all the modems will maintain good Signal-to-Noise ratio, will synchronize correctly and will maintain the communication path regardless of the multisited path in use.

The system design engineer must perform the same link examination for Modem audio as for Clear Voice and must additionally ensure that the loss has no effect on the lower level of the modem audio. If any link has excessive loss, the links must be evaluated for modification or redesign. The links should be aligned to the level required of the worst (most loss) link to ensure that the modem audio presented to the destination equipment will be at approximately the same levels across the system to maintain synchronization. Consideration must be given to margin for some increased loss over the link over time such as in Leased Lines.

4.3.2 Digital Voice Calls To A DVIU

Multisite Digital calls terminating at Conventional Station interface, Dispatch Console, Logging Recorder or Digital Interconnect utilize DVIUs. The IMC will allocate a DVIU to the call and the Base Station GETC modem will train and synchronize to the DVIU modem. The DVIU modem provides Modem audio-RS232 conversion, applies the data to the Aegis module for decryption and/or D-A conversion, and that derived Clear Voice audio is returned to the IMC for routing to the appropriate destination equipment interface. Note that (Jessica) Digital Interconnect requires two DVIUs for Interconnect calls, one for the inbound path to the Jessica and one for the outbound path. The modems will re-synchronize for each transmit and receive portion of the call.

The return path is much the same. Clear Voice is passed via the IMC to the allotted DVIU to be digitized/encrypted in the AEGIS module. The AEGIS module passes the digital voice to the Modem Interface Board which provides Modem audio- RS232 conversion and then passes the modem audio to the IMC. The IMC switches the modem audio to the appropriate Multi-Site interface for transmission to the appropriate Base Station GETC modem.

Incoming Multi-Site Digital Voice calls are applied as modem audio to the DVIU via R7 which provides the same modem input gain control as R1 on a GETC. The modem passes the derived Digital information to the AEGIS Module for conversion to Clear Voice. The DVIU provides gain to that Clear Voice via R23 on the modem interface board and returns the audio to the IMC for routing to the appropriate destination equipment.

Outbound audio from the user interface requiring digitization is passed by the IMC to the DVIU where gain is applied by R59 and R60 prior to input to the AEGIS module. The AEGIS module digital output converted to modem audio and gain is applied to the modem output level by R9 to a level acceptable for communication to the system Base Station GETC modems.

The Multi-Site link alignment requirements for Digital Voice Systems are more stringent than for Clear Voice. As any DVIU may communicate with any Base Station GETC at any site in the system, all the Base Station GETC and DVIU Modem levels must be carefully aligned to ensure that performance is maintained across ALL Multi-Site links. Additionally, the DVIU adjustments must be carefully aligned to ensure that there is no perceivable difference between Clear Voice and Digital Voice audio levels at the Conventional Station, Dispatch Console, Logging Recorder and Jessica user interfaces.

The system design engineer should perform the same examination of the link as for Clear Voice. If any link has excessive loss, the links must be evaluated for modification or redesign. The links should be aligned to the level required of the worst (most loss) link to ensure that the modem audio presented to the end equipment will be at equivalent levels.

4.3.2.1 DIGITAL DISPATCH

The voice communications between Console and IMC for Digital Dispatch is Analog Clear Voice audio and therefore the alignment requirements for Digital Dispatch do not differ from that of Clear Voice Dispatch. The Enhanced Audio Enclosure, Console and Console Interface Module (CIM) are together aligned so that incoming audio is at a specified level and all outgoing audio is set at a level acceptable to the IMC Interface.

The alignment for Digital Dispatch occurs in the DVIU, which is aligned, as in the previous section, to ensure that there is no perceivable difference between Clear Voice and Digital Voice calls when either received at the Console or heard at the radio terminal.

4.3.2.2 LOGGING RECORDER

The alignment requirements for analog Logging Recorder operation in a Digital Voice system do not differ from that of a Clear Voice system. The DVIU is aligned, as for Digital Dispatch to ensure that there is no perceivable difference between Clear Voice and Digital Voice calls as received by the Logging Recorder. The remaining alignment is between LRIM and Recorder equipment.

4.3.2.3 DIGITAL INTERCONNECT

The IMC to Jessica audio path alignment requirements for Digital and Analog Interconnect are the same. The difference from other digital operations is that two DVIUs are assigned to the Interconnect call, one to the outbound portion of the call and one to the inbound portion of the call. The DVIUs are, again, aligned to ensure that there is no perceivable difference between Clear and Digital Voice Interconnect calls.

4.3.2.4 CONVENTIONAL STATIONS

The alignment requirements for Digital Calls to and from Conventional Stations are no different from that of Digital Dispatch. The DVIU will convert digital calls to clear voice which will be directed by the IMC to the Conventional Station interface for transmission to the appropriate conventional station. the received conventional clear voice will be presented to the IMC, transferred to the allotted DVIU for conversion to digital voice and then directed to the destination equipment.

The DVIU should be aligned as for Digital Dispatch to ensure that there is no perceivable difference between Clear and Digital Voice calls terminating at the Conventional Station. The conventional and IMC interface equipment should be aligned to ensure that receive and transmit audio are at levels acceptable to the DVIU for it's normal operation.

Be aware of the requirement for Securit and Hold 2175 Hz tones, the former is 10 dB higher than average voice (i.e. at Voice Peak) and the latter is 30 dB below Securit. Ensure that the Hold tone is not lost in the link noise.

4.4 DATA SYSTEMS

Data capability may be incorporated in either Clear Voice or Digital Voice systems. The Data requirement may be for either RF Data requiring only Base Station GETC modem to Base Station GETC modem communication, or may be Landline requiring Base Station GETC modem to communicate with similar modems in the EDG. The communications paths and alignment requirements for RF Data are as for multisited Trunked Radio-to-Radio Digital calls. Those for Landline Data are defined below.

The data is passed between Site and EDG as for Digital communications, the IMC switching the modem audio to the EDG equipment. The IMC Interface for the EDG provides no controllable gain but the Modem interface module at the EDG can provide two settings for gain adjustment to the input signal by the use of a jumper. The input jumper, P3, can be set for an input level of either 0 dBm or -10 dBm. The modem audio is then passed to a modem which provides A-D conversion and the derived Data is passed to the EDG.

The Data from the EDG is passed to the modem for D-A conversion and gain is applied to the output by jumper P2 which has two settings for the desired output level of either -2 dBm or -12 dBm. This modem audio is passed to the IMC for switching to the destination Base Station GETC modem.

Again, the various multisite links must be carefully aligned to ensure that the modem levels from every site are approximately equal when presented to the EDG. This will ensure that there will be no perceivable difference in performance of Data calls between the various multisite links.

5. ALIGNMENT

5.1 SITE ALIGNMENT

5.1.1 Base Station Alignment

Refer to AE/LZT 123 3242/1 for MASTR III Base Station and GETC alignment procedures and refer to the appropriate GETC SRNs for the applicable Dip Switch Settings for each alignment test.

The fundamental requirements are that:

- the Base Station and GETC will be aligned to provide 60% of System Deviation for the **calculated** Clear Voice and Modem input from the link,
- the Base Station and GETC audio outputs will be aligned for the **calculated** input to the IMC at levels required to compensate for the losses of the link,
- if there is to be any difference between Base Station and GETC Modem audio levels, the difference is maintained for all Multisite links in the system.
- Downlink/Uplink GETC Modem transmit levels should be aligned to compensate for the losses of the link.

The nominal audio input and output levels for both Base Station and IMC is -10 dBm (0 dBm Vpk) and Base Station GETC modem output level will be 5 dB below that of Base Station Clear Voice output. The 0 dBm Vpk value allows for occasional high peaks to a value of +3 dBm Vpk without driving the Transmitter into limiting. The station should be aligned to go into limiting 3 dBm higher than Vpeak. Tx Limiting is to be set with an input level calculated to be 3 dBm higher than the Voice Peak level expected for normal operation of this channel and at a level no higher than +10 dBm.

The Base Station may be aligned for a calculated input and output level prior to connection to IMC, but should be verified once the link is in place. The calculated level in and out of the IMC may be entered into the IMC Manager’s interface configuration screen prior to connection to the Base Station but the actual measured value must be entered once the multisite link is in place.

If the input and output levels and alignments are correct but communications are still distorted, examine the specifications of the link equipment and ensure that the link levels are within audio range specifications and that the voice is not being clipped or otherwise distorted.

The following alignments (those in bold are essential for level alignment) are to be checked and realignments made when necessary levels differ from the factory defaults:

- Low Speed Data CG
- Repeater Gain RG
- Transmit Limiter TX
- **Station Line Output Level LO**
- **DSP Line Input Sensitivity DI**
- Compressor Gain CP
- Compressor Threshold CT
- **Modem Line Input R1**
- High Speed data deviation R31
- **Modem Line Output Level R2**
- SINAD
- Squelch
- Transmitter Output Power PA

5.1.2 Downlink Alignment

- **Modem Line Input R1**
- **Modem Line Output Level R2**

5.2 MULTISITE LINK

The IMC MIM module has an input/output range as follows:

- INPUT: +12 to -25 dBm Vpk
- OUTPUT: +10 to -16 dBm Vpk

The Base Station has an input/output range as follows:

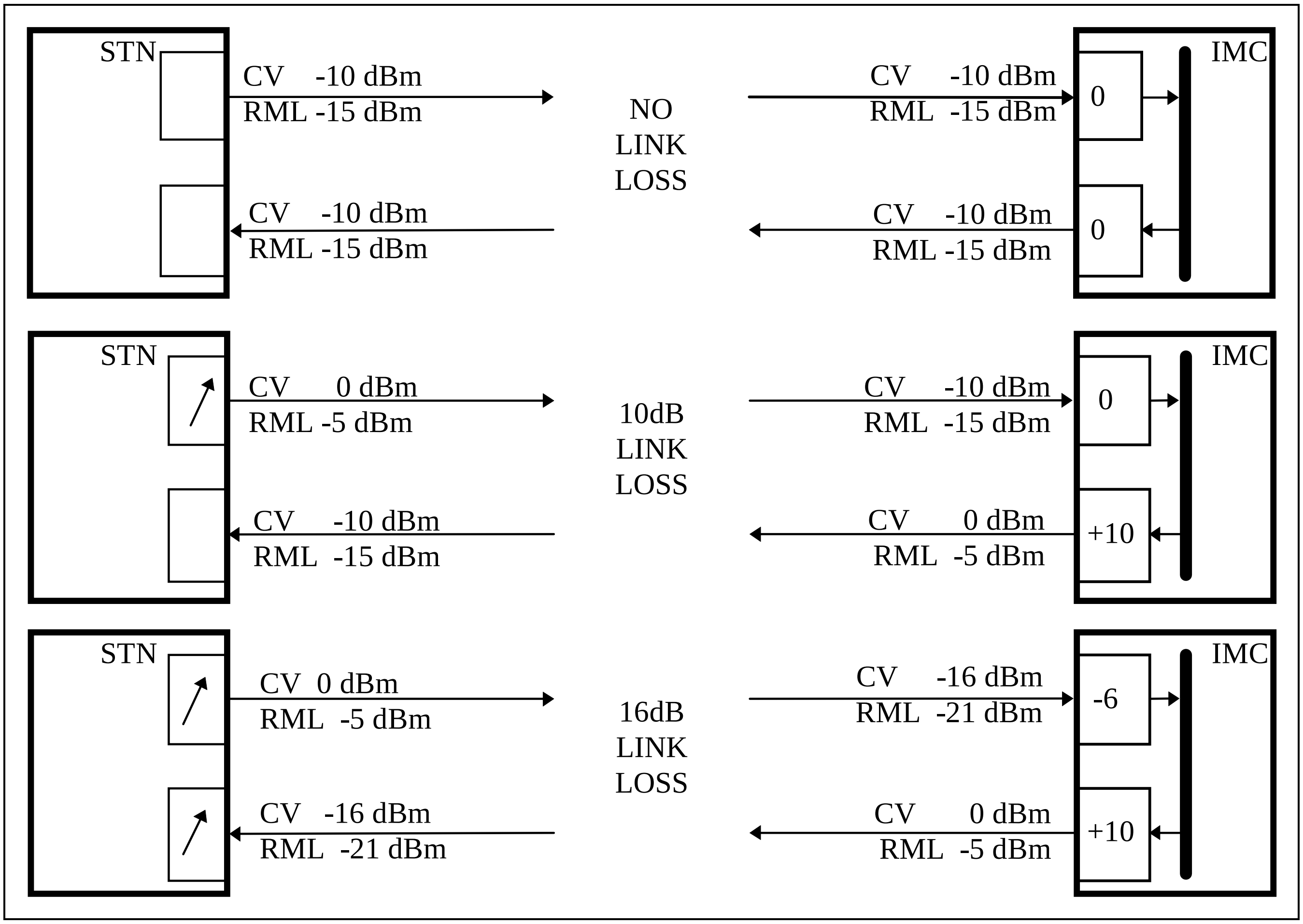
- INPUT: +11 to -20 dBm Vpk
- OUTPUT: +11 to -20 dBm Vpk

The Base Station, Uplink and Downlink GETC Modems have input/output ranges as follows:

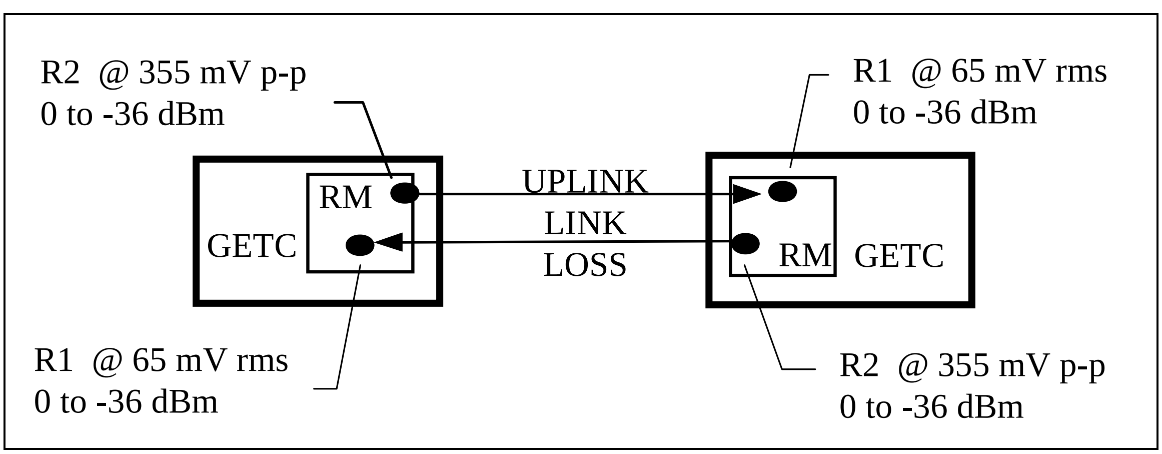
- INPUT: 0 to -36 dBm Vpk
- OUTPUT: 0 to -36 dBm Vpk

Base Station and IMC Voice and GETC Modem levels

Note: Levels are at AVTT with a designed CV level of -10 dBm and RML level of -15 dBm.



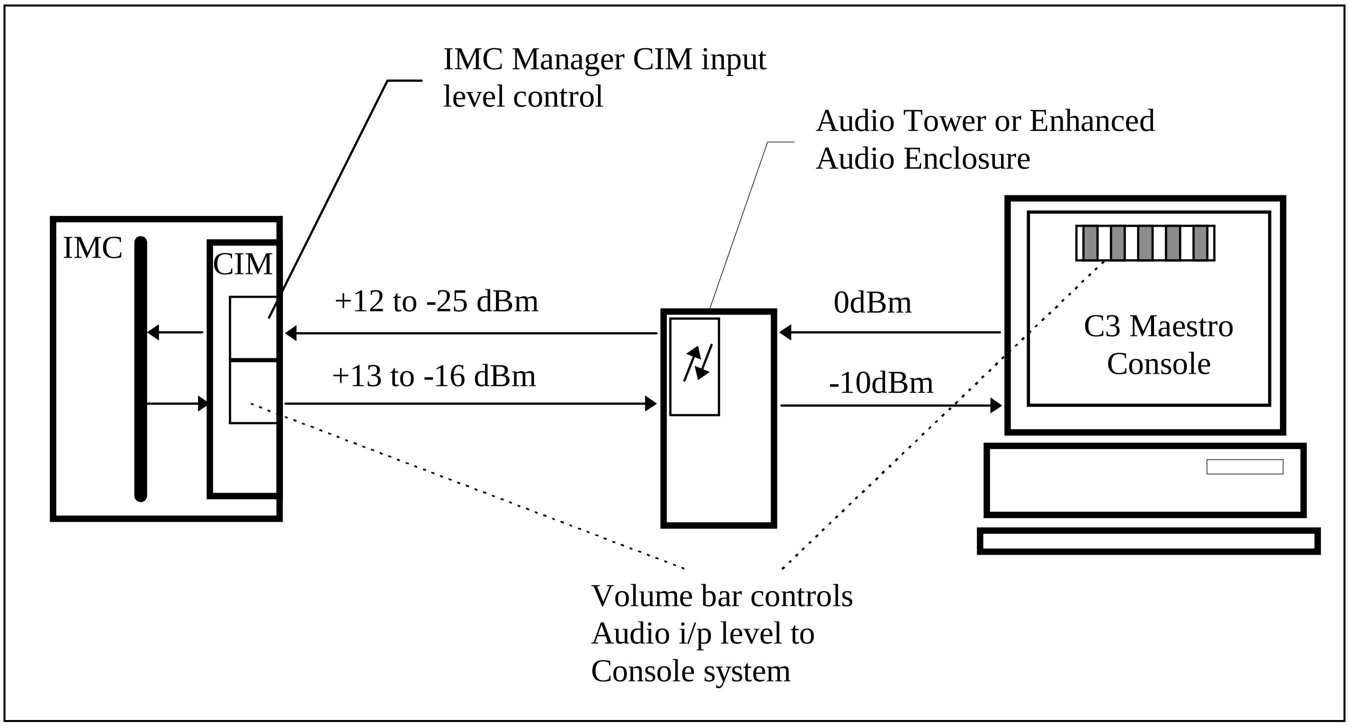
Uplink-Downlink Modem level alignment



5.2.1 DISPATCH

The output of the CIM is adjusted by the Volume Bar level setting on the Maestro. At mid-level, the input level to the Audio Tower or Enhanced Audio Enclosure is to be -10 dBm (nominal) or to the desired level. The Mic output from the Console System is to be -10 dBm (nominal) or to the desired level and that output to the IMC can be modified by the CIM level setting.

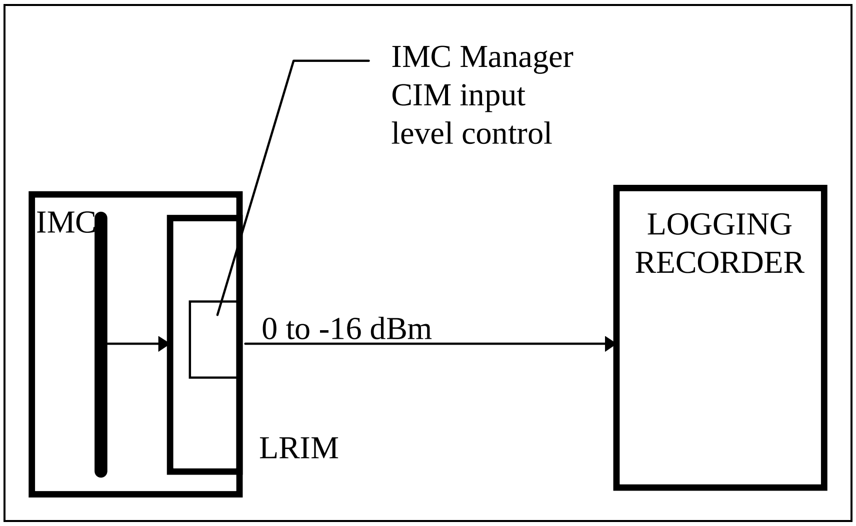
For Digital Dispatch, perform a DVIU alignment and confirm that there is no difference between Clear Voice and Digital calls both for transmit and receive audio. If there is a difference, realign the DVIU.



5.2.2 Logging Recorder

The output of the LRIM is adjusted by the IMC Manager LRIM level setting. Using the input specification for the particular Logging Recorder in use, decide what the Vpk level is required to be and set that dBm level in the IMC Manager LRIM configuration screen.

For Digital Dispatch, perform a DVIU alignment and confirm that there is no difference between Clear Voice and Digital calls both for transmit and receive audio. If there is a difference, realign the DVIU.



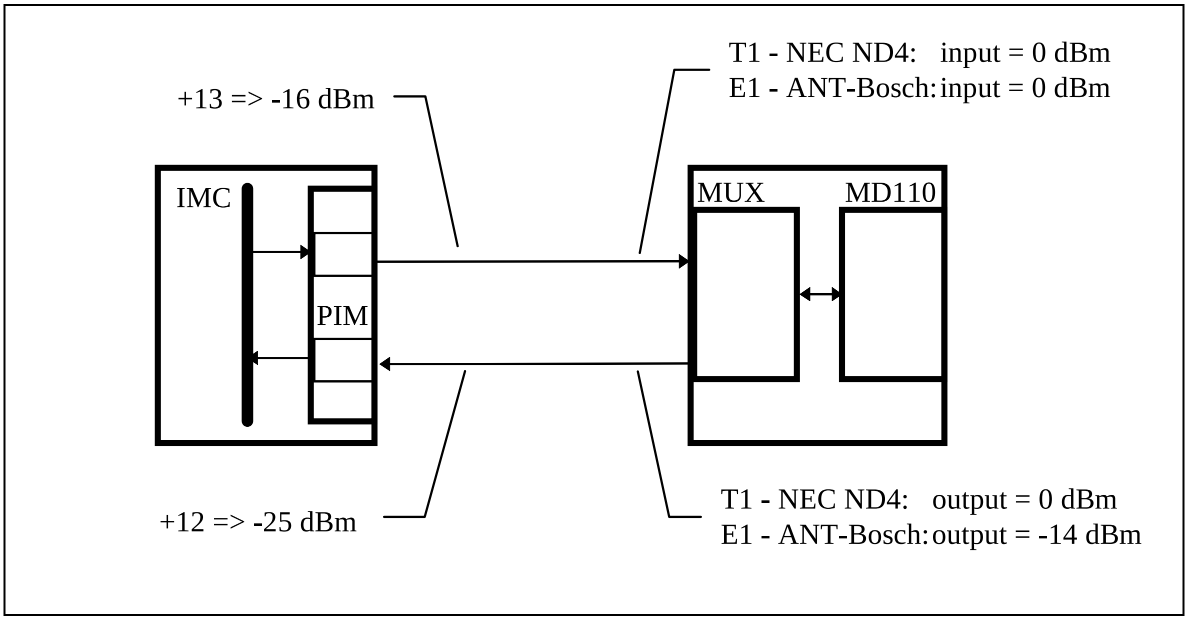
5.2.3 Jessica Interconnect

The PIM level control at the IMC Manager is in dB Voice Peak and the figure entered into the control field represents the actual level, input or output, required at that point. In the near future, all PIM Links will be T1 only and will require no adjustments.

The Jessica equipment is used for T1 and E1 applications and has different multiplexers, and therefore different input and output ranges, for each configuration the T1 multiplexer will be NEC ND4 and the E1 multiplexer will be ANT-Bosch.

The ranges are as follows:

NEC ND4:	input = 0 dBm	output = 0 dBm.
ANT-Bosch:	input = 0 dBm	output = -14 dBm.



5.2.4 DVIU Alignment

All signal levels alignments for a DVIU channel are located within the DVIU Interface Unit. There are no adjustments available for the DVIM from the IMC Manager. Adjustments include three (3) potentiometers on the Interface Board and two (2) potentiometers on the Tone Remote Control Board. A signal path block diagram is located at LBI-39043.

The alignment described below assumes that the rest of the input and output modem levels in the EDACS System have been aligned for a nominal value of -10 dBm. If the levels in the customer's system differ, make the appropriate changes to the levels shown below.

Since each DVIU channel is independently adjustable, these alignment procedures are to be repeated for each DVIU channel in the system.

5.2.4.1 SETUP

1. If the DVIU Channel is pooled, temporarily disable all other DVIU Channels via the IMC Manager. If the alignment is being made on a live system allocate the DVIU to a test group and perform all tests on that group.
2. Connect a calibrated TTS to the DVIM output at the punchblock with bridging clips removed, or disconnect DVIU input cable at TB1 and connect TTS to TB1 input cable pins 1 and 2, set the TTS to 600Ω termination mode, and confirm DVIM output is at -10 dBm. Reconnect cable and/or bridging clips and remove TTS.

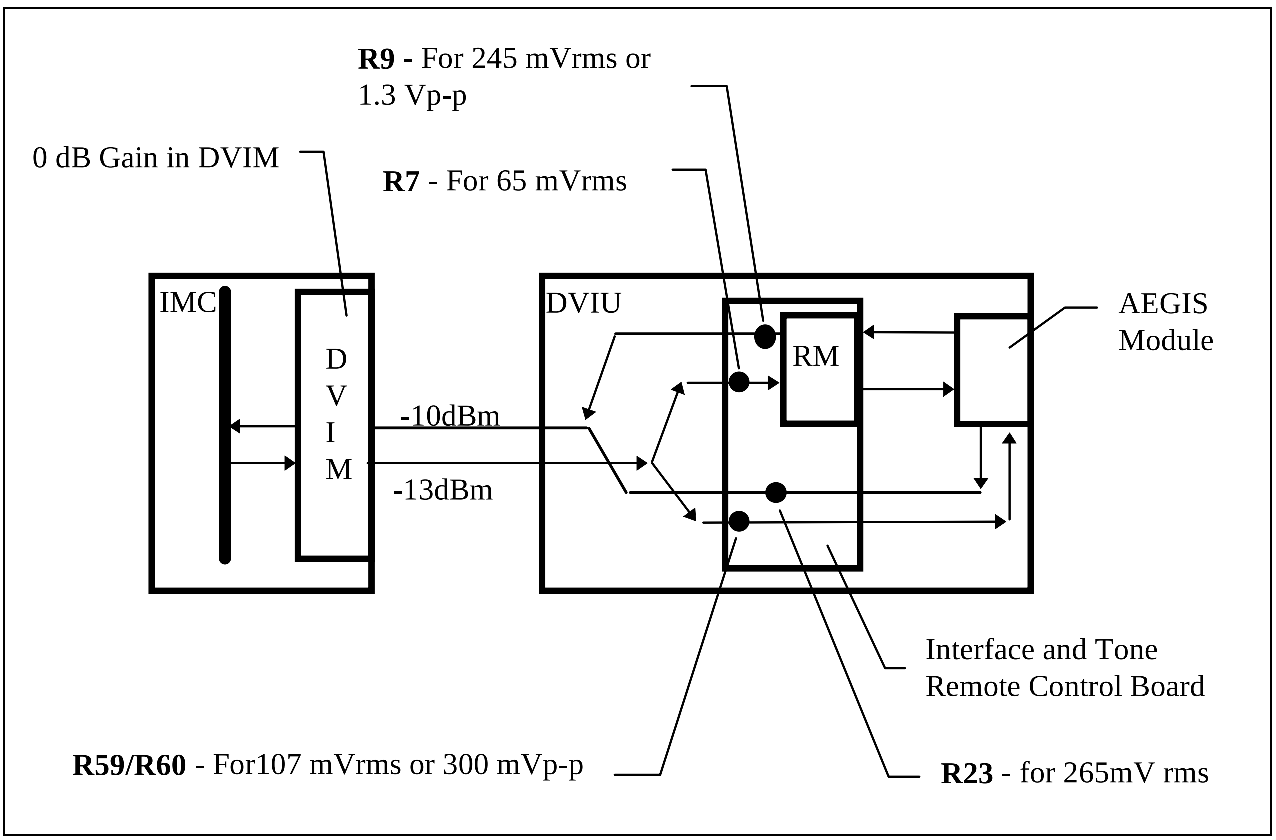
Note: When the DVIU is connected to the DVIM, the DVIU input level is attenuated by 3 dBm to -13 dBm.

5.2.4.2 RADIO-ORIGINATED CALL

This adjusts the Digital Voice encoded/encrypted call input level to the Rockwell Modem.

1. Connect True Vrms Voltmeter, to Interface Board TP1.
2. Set test radio to DV Group and PTT.
3. Adjust **R7** for 65 mVrms (-22 dBm) at TP1.
4. Unkey radio.

This adjusts the decoded/decrypted call Clear Voice output level from the Aegis module.



5. Connect the TTS across DVIU TB1 pins 3 and 4.

6. Switch the test radio to CV Mode, PTT and inject a 1 kHz tone to the Mic at a level to produce 60% system deviation. Switch the radio to DV mode with the tone injected and PTT.

7. Adjust **R23** for 265 mVrms at TB1.

8. Unkey radio.

5.2.4.3 CONSOLE/JESSICA-ORIGINATED CALL

This adjusts the Clear Voice non-encrypted call input level to the Aegis module.

1. Disconnect TB1 to isolate DVIU and jumper TB1 pins 1 to 8 and 2 to 9.

2. Connect True Vrms Voltmeter across J6 pins 1 and 2 (Mic Hi and Lo).

3. Connect audio generator across TB1 pins 1 and 2 and inject a 1 kHz tone at a level of -13 dBm.

4. Adjust **R59** on Tone Remote Board for Mid-level.

5. Adjust **R60** on Tone Remote Board for 107 mVrms (300 mVp-p).

This adjusts the encoded/encrypted Digital Voice call output level from the Rockwell Modem.

6. Connect True Vrms Voltmeter across TB1 pins 3 and 4.

7. Short DVIM M Lead by jumping TB1 pins 10 and 11. This places DVIU in transmit mode.

8. Adjust R9 on the Interface Board for 245 mVrms at TB1.

9. Disconnect all jumpers, re-connect DVIU TB1 and remove Test Equipment.

5.2.4.4 SYSTEM TEST

Place a DV call on a DV group from a radio and observe the CV audio level at the Console/Jessica/Logging Recorder. Place a CV call on a CV group from the radio and observe the CV audio level at the Console/Jessica/Logging Recorder. The audio levels should be the same.

Place a DV call on a DV group from the Console and observe the DV audio level at the radio. Place a CV call on a CV group from the Console and observe CV audio level at the radio. The audio levels should be the same.

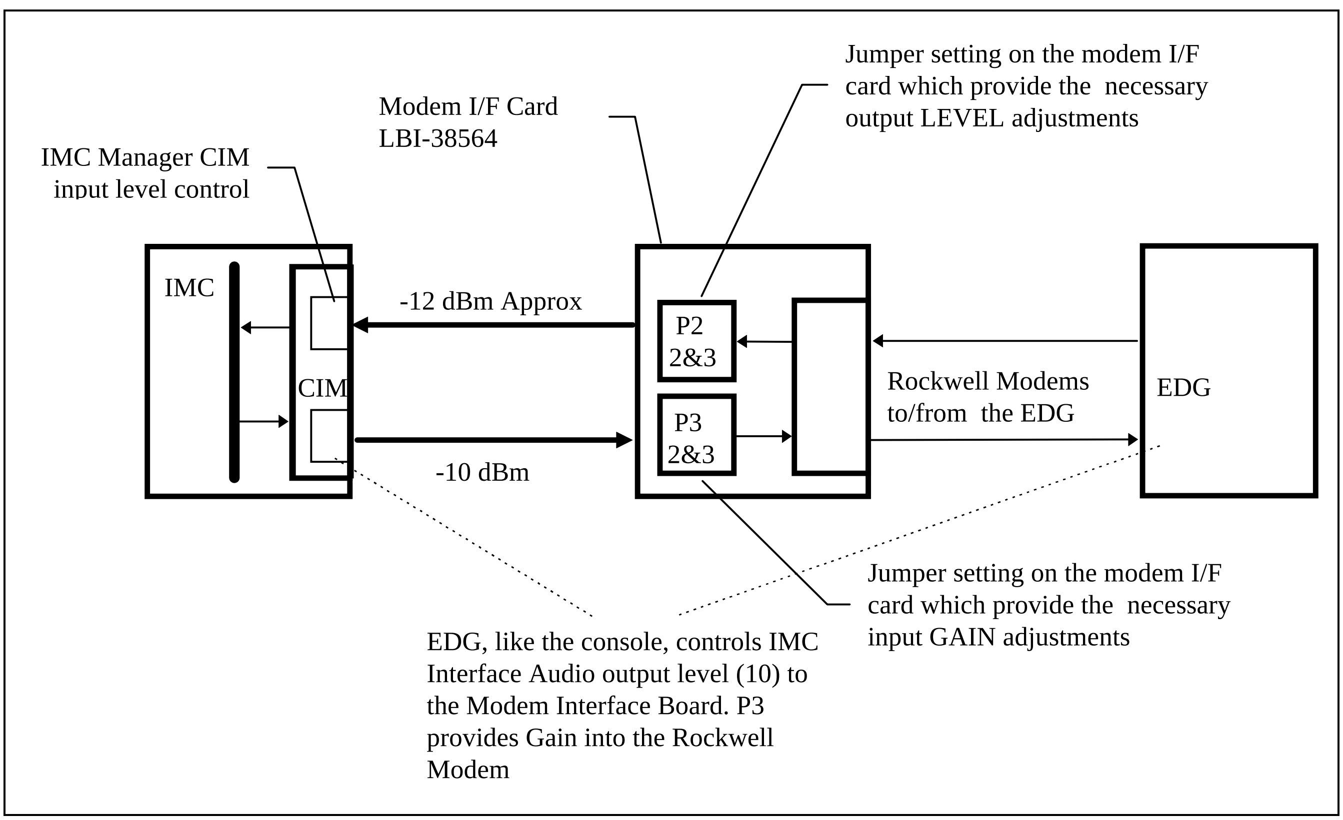
5.2.5 Data

The Interface Module at the IMC for the DATA Link is a CIM and, as such, controls the audio output level of it's IMC Interface Module. There are no IMC Manager Level Controls for the DATA Link. The EDG input level control is performed by both the EDG controlling the CIM (equivalent to the Console Volume Bar) and the EDG Modem Interface Card gain setting jumpers. The EDG output level is set by the EDG Modem Interface Card gain setting jumpers.

The settings shown below provide for system modem audio requirement of -10 dBm.

The jumper P3 performs the same function as GETC R1 in that it provides two levels of gain to the incoming modem audio. Position 1&2 is for an input gain of 0 dBm, 2&3 is for an input gain of -10 dBm.

The jumper P2 performs the same function as GETC R2 in that it provides two levels of gain to the output modem audio to the CIM. provides output level control for the Modem Interface Board. Position 1&2 is for an output level of -2 dBm, 2&3 is for -12 dBm.



6. GLOSSARY OF DEFINITIONS

AVTT: **Average Voice Test Tone** is defined as a 1 kHz tone at the level **Average** Voice will be input to the station. For equipment aligned in the factory, this level will be -10 dBm (245 mVrms) producing nominal deviation at the transmitter of 60% rated system deviation. This allows for voice peaks at a level 10 dB higher, Hold tone at a level 20 dB lower and Base Station Transmitter Limiting at +3 dBm.

AVTT at -10 dBm = Voice Peak at 0 dBm.

TEC Interface: The TEC Interface is an IMC Link equipped with T1/E1 Interface Card. The only level control adjustments available for links equipped with TECs is at the MUX terminating the Link. Note that this AVTT and RML level is acceptable to the input and output range of the Intraplex Mux. Note: Mux specifications refer to Voice Peaks. Input and output ranges of other vendor Mux equipment must be examined for requirement.

The equipment connected to the multiplexer equipment will nominally be aligned for -10 dBm AVTT & -15 dBm RML in and out of the Link. If the link has no Mux, i.e.; NIM Link, then there will be NO level control. The TEC Interface IMC does not use Subaudible tones and therefore the Mux input range does not need to allow for levels 20 dB below AVTT.

Transmitter Deviation: This means 60% rated system deviation for the type of system in use. This document will utilize deviation settings for **Wide Band systems**. For systems other than Wide Band, refer to the table below for the equivalent settings:

	System Deviation	Low Speed Data	Tx Limiter
Wide Band	3.0 kHz	750 Hz	4.5 kHz
NPSPAC	2.4 kHz	600 Hz	3.0 kHz
Narrow Band	1.5 kHz	500 Hz	2.25 kHz

Signal Levels: Levels are shown in either Vrms (True), Peak to Peak or dBm (Average Responding RMS Calibrated Transmission Test Set (TTS or TIMMS)).

SIGNAL TYPE	LEVEL	PEAK - PEAK	V RMS
Voice Peak	0 dBm	2.17 V	775 mV
AVTT	-10 dBm Sine	686 mV	245 mV
RML	-10 dBm Modem	1.3 V	265 mV
RML	-22 dBm Modem	355 mV	65 mV

Rockwell Modem Level: RML O/P: Rockwell Modem line output is to be adjusted at R2 for 5 dB below that of Clear Voice as measured using an Average Responding RMS Calibrated Transmission Test Set at punchblock or equivalent station output demarcation point. This allows for RML peaks 5 dB higher than the level read by an Average Responding Voltmeter Test Set.

RML I/P: Rockwell Modem Input is to be adjusted at R1 for 355 mV p-p (65 mVrms) as measured at U18-A Pin 1. This assumes -10 dBm input to the punchblock or equivalent station input demarcation point.

Station High Speed Data: Align GETC R31 to set Station HSD. The data derived from the Rockwell Modem I/P is passed to the GETC Logic board where it is routed to the HSD input to the station. R31 sets the transmitter deviation due to HSD.

Secur-it Tone: A 2175 Hz tone at 10 dB above AVTT used to key the station from Console or IMC. This will be at Voice Peak.

Hold Tone: A 2175 Hz tone at 20 dB below AVTT (30 dB below Vpk) used to hold the station keyed for the duration of the call.

Link Loss This is the Loss of a Link expressed in dB (16 dB is signal attenuation by 16 dB)

IMC Manager Level Control: The IMC Manager level control screens require the actual peak level of the input and output signal to be entered in the level control field. This level is expressed in terms of Voice Peak, not a dBm level, and will therefore be 10 dB higher than that (AVTT) suggested in this document.

$\boxed{0}$ This symbol indicates the level control setting in the appropriate IMC Manager screen which will adjust the input/output level for a link. A zero here indicates and input or output level of 0 dBm Vpeak or -10 dBm AVTT

$\boxed{\nearrow}$ This symbol indicates gain provided by the level control.

Vpk Voice Peak

TELCO LINE TYPES and DEFINITIONS:

DECIBEL (dB) - A unit of measurement representing the logarithmic a ratio of two voltages, currents or power levels; used in telecommunications to express transmission loss or gain; defined as one-tenth of a Bel, hence the appropriate notation is dB, shown here.

dBm - Identifier meaning "decibels referred to one milliwatt," the common reference point for power levels in telecommunications circuits.

dBm0 - Identifier meaning "decibels referred to one milliwatt and corrected to a Zero dBm effective power level;" used to state the relation of a signal level on a transmission line at other than a one-milliwatt point.
Example: Throughout an analog system, a data set signal is to be kept 13 dB below that for a single test tone, stated as "minus 13 dBm0;" at a carrier modulator input where test tone level is -16 dBm, a data signal should then be 13 dB lower, or -19 dBm.

dBmp - Identifier meaning "decibels below reference tone using psophometric (filter) weighting," the CCITT method for noise measurements; has about 2 dB variance from Bell methods.

dBm0p Identifier for CCITT psophometric-weighted noise measurements adjusted to a relative 0 dBm transmission level point.
Example: An absolute measurement of minus 40 dBmp noise at a carrier channel output point would mean a signal-to-noise ratio of about 47 dBm0p exists at that point on the circuit.

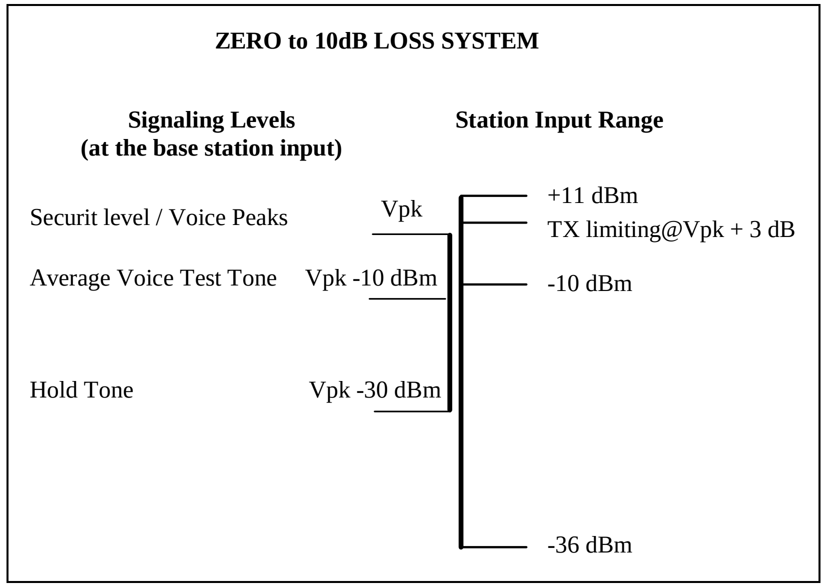
dBrn - Identifier meaning "decibels above reference noise," the reference commonly used being 90 decibels below one milliwatt.

dBrnC - Identifier meaning "decibels above reference noise measured through a (filter) weighting network approximating a " type C voice message channel;" the common North American nomenclature for a DDD trunk channel; having a reference of 90 decibels below one milliwatt of power.

dBrnC0 - Commonly pronounced "de-brink-o," identifier meaning "decibels above reference noise with C message weighting adjusted for equivalence to a 0 dBm (one milliwatt) equivalent circuit point."
Example: A direct measurement of 49 dBrnC0 at the nominal +7 dBm output of a carrier demodulator would mean the noise had been offset by 7 dB; thus the reading in dBrnC0 would be 42.

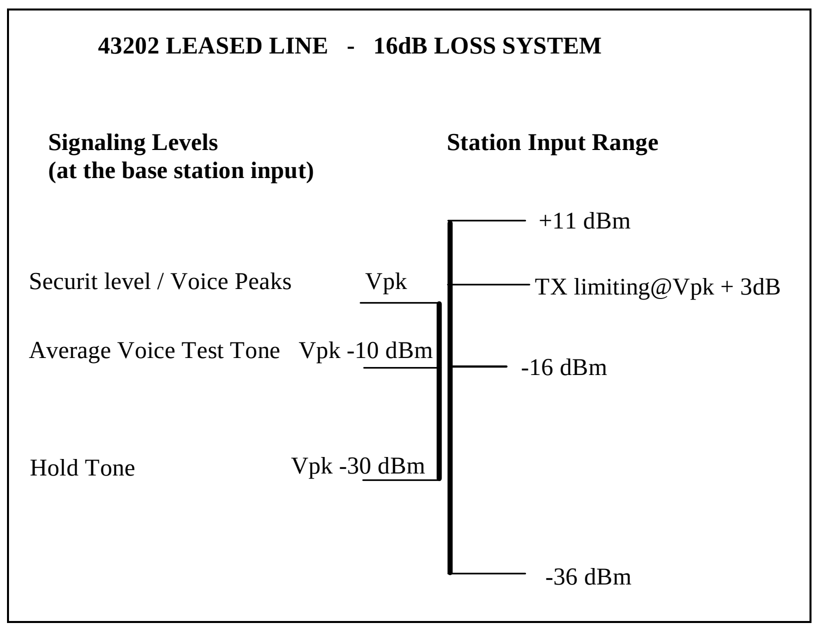
7. APPENDIX A - BASE STATION INPUT/OUTPUT LEVELS

When selecting levels at which to operate the base stations in the system, one should bear in mind the optimum level of -10 dBm for AVTT and of -15 dBm for RML and the net amount of link loss between the Remote audio signal source and the base station (see figure 1). If the net loss between the station and the remote source is 0 dB or no loss, then it is desirable to set up the station for -10 dBm (AVTT) and TX limiting at +3 dBm. If there is some amount of loss between the two, 10 dB or less, then the inputs and outputs of the linked equipment should be increased to maintain the level of -10 dBm (AVTT) at the station input.



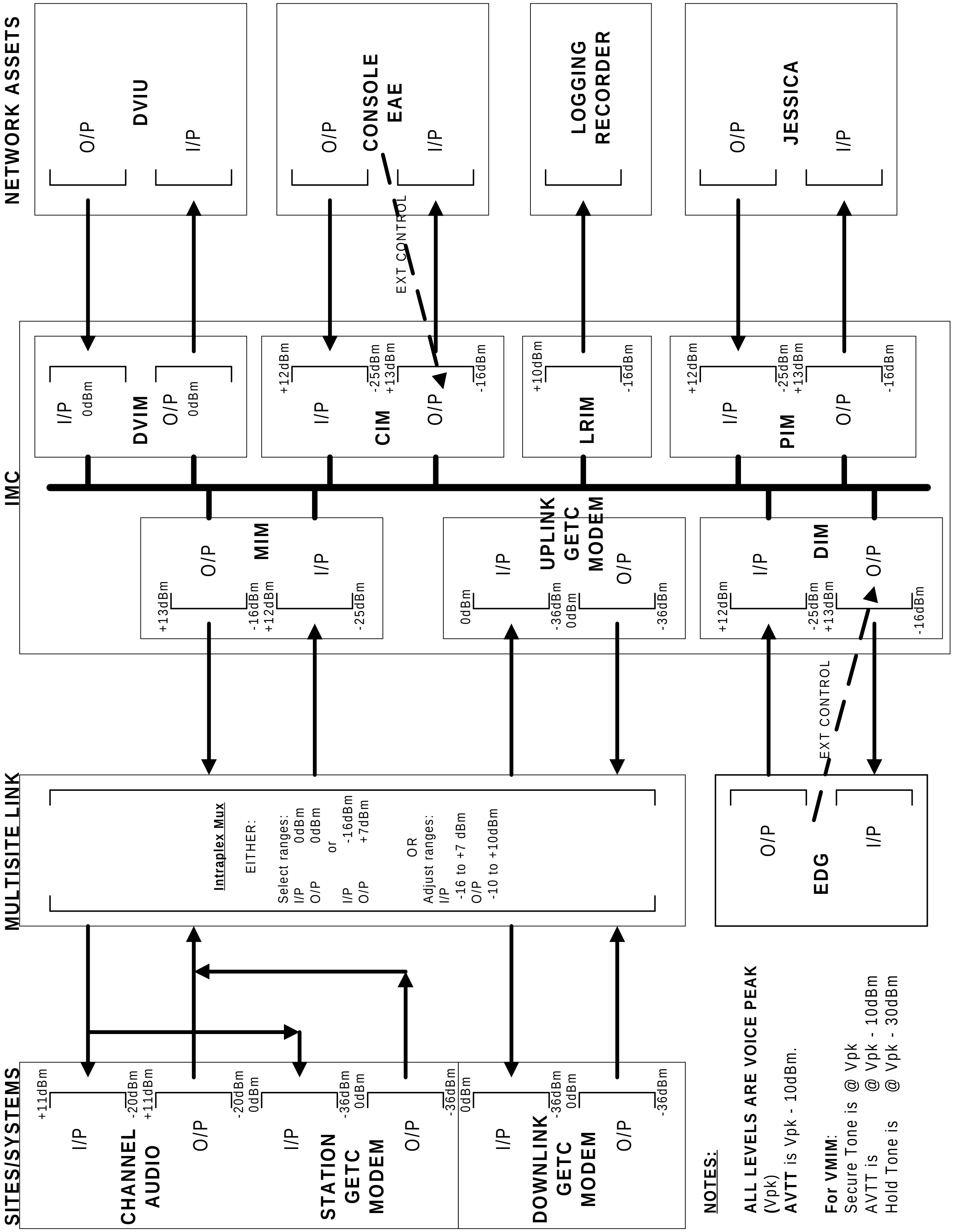
Link loss greater than 10 dB - but no more than 16 dB - cannot be compensated for solely by the source equipment. The balance of loss over 10 dB should be compensated for by both increasing the levels into the link to the station AND by lowering the value of required input (AVTT & RML) to drive the station to 60% system deviation whilst maintaining a good Signal-to-Noise Ratio.

For example, In typical phone company leased lines, the loss is rated at 16 dB maximum. In this case of 16 dB of link loss, to maintain an acceptable Signal-to-Noise Ratio and to achieve 60% system deviation for both AVTT and RML, the IMC output should be set to the maximum allowed for the link. The residual loss will be compensated for by the Base Station and GETC input level controls. The reverse will be true for Base Station and GETC output to the IMC.



There should be a maximum of 16 dB loss allowed in an EDACS Link. Should the Link have a greater loss, the customer must be alerted to this fact and it must be understood that the link must be upgraded or replaced.

8. APPENDIX B - EDACS SYSTEM INPUT/OUTPUT LEVELS



9. APPENDIX C - LEASED ANALOG CIRCUIT SPECIFICATIONS AND TEST PROCEDURES

9.1 BASIC SPECIFICATIONS:

Most circuits ordered for data transmissions over analog facilities must meet the following minimum specifications:

Type of Facilities:	Four-wire facilities, one pair for transmit and one pair for receive, terminated on a USOC registered jack (usually RJ-11).	
Impedance:	600 ohms resistive balanced	
Transmit level:	0 dBm at demarc block	
Receive level:	-16 dBm, at 1000 Hz, at demarc block	
Frequency response (ref 1000 Hz):	300-2500 Hz, -3 to +12 dB	
	500-2500 Hz, -2 to + 8 dB	
Frequency Error:	+/- 5 Hz	
Delay characteristics:	Absolute delay	not specified
	Envelope delay	<1750 microseconds from 800-2500 Hz, ref 1600 Hz
Noise characteristics:	Signal to noise ratio	24 dB minimum
	Impulse noise	not more than 15 counts in 15 minutes at -24 dB threshold (measured at -21, -24 and -27 dBm0)
Conditioning Options:	D type conditioning removes companders, line taps and load coils from all circuit elements. In addition the noise specifications are improved as follows:	
	Signal to noise ratio	28 dB minimum
	Impulse noise	not more than 5 counts in 15 minutes at -28 dB threshold (measured at -25, -28 and -31 dBm0)

C2, C4 and C5 (or D2, D4 and D5) conditioning improves the delay characteristics of the circuit in stages with the best improvement in the higher numbers (costs more). This is usually not necessary, as most modems offer adaptive equalization in their design. If external equalization is required, the circuit will need to be modified to conform to specifications included in the tariff's specification (at an increased cost to the user).

9.2 LEASED DATA CIRCUIT TEST FORM

Date:______________________________ Circuit Number:______________________________

EDACS Channel Number:______________________________

Terminal Location 1:	____________________ Site “A”	____________________ Technician Name
Terminal Location 2:	____________________ Site “B”	____________________ Technician Name

Test equipment required: HP 4934A TIMS (or equivalent) ______________________________

Specify Type (a)

9.3 AMPLITUDE TESTS:

Perform these tests in both directions record readings below: (Exact readings please)

1. Insert 1004 Hz, 0 dBm at 600 ohms, to transmit pair at demarc block, site A. Read frequency and level terminated in 600 ohms at receive pair on demarc block at site B.

Readings_________________________Hz, ________________________ dBm.
Should be 1004 =/- 5 Hz Should be -16 dBm

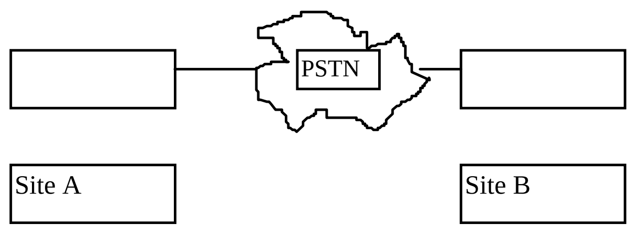
2. Insert 1004 Hz, 0 dBm at 600 ohms, to transmit pair at demarc block, site B. Read frequency and level terminated in 600 ohms at receive pair on demarc block at site A.

Readings_________________________Hz, ________________________ dBm.
Should be 1004 =/- 5 Hz Should be -16 dBm

3. While equipment is already connected in the B to A direction, change send frequency to 404 Hz at site B. Observe the frequency and level at site A. Change frequency in 200 Hz steps at site B, recording readings observed at site A for each step in table below:

Rev Level Readings

Freq	Site A	Site B	
404	_____	_____	
604	_____	_____	
804	_____	_____	
1004	____	_____	Should be -16 dBm
1204	_____	_____	
1404	_____	_____	
1604	_____	_____	
1804	_____	_____	
2004	_____	_____	
2204	_____	_____	
2404	_____	_____	
2604	_____	_____	
2804	_____	_____	
3004	_____	_____	
3204	_____	_____	
3404	_____	_____	
3604	_____	_____	Observe these readings to determine if facility is over a loaded cable.
3804	_____	_____	



4. Reverse transmission direction from A to B as in step one above. Change send frequency to 404 Hz at site A. Observe the frequency and level at site B. Record amplitude readings observed at site B above. Change frequency in 200 Hz steps at site A, record readings observed at site B for each step in table above.

9.4 NOISE MEASUREMENTS:

1. With test equipment connected as in step 4 above, return send frequency to 1004 Hz at site A. Set test set at site B for “Notched Noise Test” (refer to mfg. documentation for test information and set-up instructions - may vary from test set to test set). Observe reading at site B; and record reading below.

Notched noise test: 1.________________ 2.________________ dBmC
Site B reading Site A reading

This test measures all noise in the channel in the presence of a test tone. Readings are in positive units of dB relative to -90 dBm and are C message weighted.

2. Reverse transmission direction. Set site B send frequency to 1004 Hz. Set test set at site A for “Notched Noise Test”. Observe reading at site A and record reading above.

3. Remove test tone from site B. Measure idle channel noise in both directions and record readings below:

Idle noise test: 3.________________ ________________ dBmC
Site A reading Site B reading

4. Reconnect test equipment to circuit as in step 1 above. Set site B for “Impulse Noise Test” (refer to mfg. documentation for test information and set-up instructions - may vary from test set to test set). Set for 15 minute test, -40 dBm (-24 dBm0) threshold. Start test and do not disturb for 15 minutes. Observe counter results and record below.

Impulse Noise Test:

Direction A to B: 4. __________, __________, __________ counts.
-21 dBm0 -24 dBm0 -27 dBm0

Direction B to A: 5. __________, __________, __________ counts.
-21 dBm0 -24 dBm0 -27 dBm0

5. Reconnect test equipment to circuit as in step 2 above. Set site A for “Impulse Noise Test”. Set for 15 minute test, -40 dBm (-24 dBm0) threshold. Start test and do not disturb for 15 minutes. Observe counter results and record above.

9.5 BRIDGE TAP AND LOADING COIL DETECTION TEST:

1. Duplicate the test set-ups in steps 3 and 4 above under "amplitude tests". Instead of using 200 Hz steps, use smaller steps or perform a "slow sweep" by manually adjusting the send frequency in the range from 404 to 3404 Hz. Observe the received signal level at the opposite end of the circuit for any sudden change in received level. No recorded readings are required unless drop-outs are found, then record frequency and the depth of the notch at which the drop-out occurred.

2. Note the readings in the Amplitude Tests table on page 1 at 3200, 3400, 3600 and 3800 Hz. If readings up to 3200 Hz are relatively flat (not much amplitude variation as the frequency increases), and the readings take a dramatic drop at or near 3400 Hz, this is an indication that there are loading coils associated with the circuit. In some cases, modems will be unable to operate in the presence of these devices. Note below if this test was performed in each transmission direction.

Tested for Bridge Taps: __________ Loading Coils: __________
(Y/N) (Y/N)

Direction A to B Freq of notch __________Hz, Notch depth __________dB.
Direction B to A Freq of notch __________Hz, Notch depth __________dB.

9.6 CARRIER LEVEL TEST:

Data circuits are quite sensitive to how they are handled in carrier facilities. Some FDM carriers will introduce harmonics and noise if data is transmitted at voice levels. Undesirable by-products mean degraded operation can be expected. Most leased circuits are carried over FDM and PCM facilities at a maximum transmission level 13 dB below the normal voice test level point. This test will disclose the existence of improper padding and level coordination within the circuit. The Telco supplier MUST correct this condition before we can accept this circuit.

1. Connect test equipment as in step 1 of amplitude tests on page 1.

2. Insert 1004 Hz, 0 dBm at 600 ohms, to transmit pair at demarc block, site A. Read frequency and level terminated in 600 ohms at receive pair on demarc block at site B. Reading should be -16 dBm0 at 1004 Hz.

3. Increase the transmit level at site A. Observe the corresponding increase in level of the received signal at site B. The level should increase in exact steps end-to-end. If all is well with the Telco facilities, the signal will continue to increase until the send level at site A reaches +13 dBm. The received signal at site B will be -3 dBm. If this condition is not met, Telco must correct levels in their facilities before we can accept his circuit.

4. Repeat step 3 above for the other direction of transmission. Site B sends and site A receives.

   Indicate below if this test was performed and circuit passed:

Carrier level test performed __________, passed: __________.
(Y/N) (Y/N)

9.7 PEAK/AVERAGE RATIO TEST:

Most modern TIMS sets such as the HP 4934A have a PAR test. It can disclose the existence of noise, harmonic products and other types of amplitude distortion and modulation by-products. The test is performed just like the amplitude test, steps 1 and 2. The sending end is set to send PAR tones, the receiving end is set to read the results.

The circuit is considered acceptable if the PAR test reading is 80 to 100 (100 being the best possible reading). Some sets may read a few numbers over, but you can ignore anything above 100.

The circuit is considered marginal if the PAR test reading is 50 to 79. Most high speed modems will not work over this circuit.

The circuit has failed if the PAR result is 49 or less. Corrective action is required. At least one of the previous tests should have also failed. In any case the Telco supplier should correct the condition before we can accept the circuit.

Perform the test in each direction of transmission. Record the results below:

Direction A to B: PAR __________ units.
Direction B to A: PAR __________ units.

Notes:

All tests can also be performed over existing corporate network facilities to detect and troubleshoot problems with data circuits operating over analog circuits. Where corporate facilities are in tandem with leased facilities, each segment must be considered and tested separately.

Circuits with conditioning options may require additional tests to verify delay characteristics. The test equipment specified cannot perform these tests. Specialized equipment may be rented, if required.

10. APPENDIX D - DIGITAL DATA RATES AND HIERARCHIES:

Tables 1 and 2 shown below contain a list of the standard asynchronous digital signal hierarchies, and a few important characteristics. *Tables 3 and 4* contain a similar list of the standard synchronous digital signal hierarchies.

Table 1 - North American Asynchronous Hierarchy

Signal Designation	Bit Rate (kb/s)	Number of VF Channels	Line Code	Interface Impedance	Comments
DS0	64	1	AMI (data)	135 Ω	Twisted Pair
DS1 **(T1)**	1,544	24	B8ZS/AMI	100 Ω	Twisted Pair
DS2	6,312	96	B6ZS	110 Ω	Twisted Pair
DS3	44,736	672	B3ZS	75 Ω	Coaxial Cable

Table 2 - European Asynchronous Hierarchy

Signal Designation	Bit Rate (kb/s)	Number of VF Channels	Line Code	Interface Impedance	Comments
E0	64	1	AMI (data)	120 Ω	Twisted Pair
E1	2,048	30	HDB3	75 Ω/120 Ω	Coax/Twisted Pair
E2	8,448	120	HDB3	75 Ω	Coaxial Cable
E3	34,368	480	HDB3	75 Ω	Coaxial Cable
E4	139,264	1,920	CMI	75 Ω	Coaxial Cable

Table 3 - North American Synchronous Hierarchy

Signal Designation	Bit Rate (kb/s)	Equivalent DS0s	Equivalent DS1s	Equivalent DS3s	Comments
VT 1.5	1.728	24	1		
STS-1	51.84	672	28	1	
STS-3	155.52	2,016	84	3	
STS-12	622.08	8,064	336	12	
STS-N	N x 51.84	N x 672	N x 28	N	

Table 4 - European Synchronous Hierarchy

Signal Designation	Bit Rate (kb/s)	Equivalent E0s	Equivalent E1s	Equivalent E3s	Comments
VT2	2.304	30	1		
STM-1	155.52	1,920	64	4	
STM-4	622.08	7,680	256	16	
STM-N	N x 155.52	N x 1,920	N x 64	N	

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1-800-592-7711 (Outside USA, 804-592-7711)

AE/LZB 119 1915/1 R1A
Printed in U.S.A.