

Voice Parameters and Tuning Guide for the IGX 8400, VISM, 3810, FastPAD, and VNS

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Introduction

This document describes how to tune a Cisco voice network using IGX 8400 series switches and switch software releases 8.2.5x and later.

Prerequisites

Requirements

The reader should be familiar with configuring Cisco equipment and basic concepts such as:

- circuit lines connect to voice input devices such as PBXs
- packet lines are the interconnecting trunks between the IGX 8400 series switch

Components Used

The information in this document is based on these software and hardware versions:

- IGX 8400 series switch CVM and UVM cards in networks using switch software 8.2.5x or later
- MGX 8850 series edge switch VISM card release 1.5.04
- FastPAD Multimedia device using software 8.0.1 or later
- MC3810
- Voice Network Switching

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

For more information on document conventions, refer to the Cisco Technical Tips Conventions.

Terms and Acronyms

- **AAL1** ATM Adaptation Layer 1. AAL1 supports connection-oriented Continuous Bit Rate (CBR) voice and video. AAL1 is commonly used for circuit emulation service transmission over ATM networks.
- **AAL2** ATM Adaptation Layer 2. AAL2 supports connection-oriented Variable Bit Rate (VBR) packetized voice and video. AAL2 does not have Convergence or Segmentation and Reassembly (SAR) sublayers.
- **AAL5** ATM Adaptation Layer 5. AAL5 is a Simple Efficient Adaptation Layer (SEAL). The common part AAL5 supports both connection-oriented and connectionless Variable Bit Rate (VBR) traffic.
- **ADPCM** Adaptive Differential Pulse Code Modulation.
- **ATM** Asynchronous Transfer Mode. A connection-oriented protocol for the transmission of voice, data and video using fixed-length cells of 53 octets. The rate of the cells is not periodic, hence the protocol is asynchronous.
- **CAS** Channel Associated Signaling. A method of signaling that allows PBXs or channel banks to communicate with each other. CAS is implemented by transmitting the signaling required for a single channel of traffic in the channel itself or in a signaling channel permanently associated with it.
- **CCS** Common Channel Signaling. Like CAS, CCS is a method of communicating between PBXs or channel banks. It is more sophisticated than CAS in that it uses a feature rich High-level Data Link

Control (HDLC) framed protocol for signaling like Q.931, DPNSS, or QSIG. CCS is implemented by transmitting the signaling information for multiple channels of traffic over one or two timeslots (typically timeslot 16).

- **Call Agent External** call control element also known as a Media Gateway Controller. Monitors resources of the overall system and maintains control of all connections. The Cisco VSC 3000 is a Call Agent.
- **Class 5** Class 5 refers to a type of switch used in the PSTN to provide local services to the end user. This switch connects end-users to the PSTN and provides custom features such as call waiting and three-way calling. Examples of these switches include the Lucent 5ESS and the Nortel DMS100.
- **Convergence** The amount of time needed to make a working mathematical model of an incoming speech sample.
- **CS-ACELP** Conjugate Structure Algebraic Code Excited Linear Prediction.
- **DASS2** Digital Access Signaling System Number 2.
- **dBm** Power level in decibels relative to 1 milliWatt.
- **DID** Direct Inward Dialing. Calls can be dialed from a telephone connected to an extension on a PBX to the public network without going through an operator.
- **DOD** Direct Outward Dialing. Calls can be dialed from a telephone connected to the public network directly to extensions on a PBX without going through an operator.
- **Double Talk** The situation where parties at both ends of a conference are speaking simultaneously. A quality echo canceller will provide a continuous speech path in both directions during double-talk.
- **DPNSS** Digital Private Network Signaling System Number 1.
- **DS-0** Digital Signal level 0. Part of the North American transmission hierarchy, transmitting at 64 kbps. A DS-0 is one DS-1 timeslot.
- **DTMF** Dual Tone Multifrequency. Generic name for push-button telephone signaling that uses two tones to represent each digit on the telephone keypad. The tones are in two groups within the speech band, a low band and a high band. They are geometrically spaced to ensure that any two frequencies of a valid combination are not harmonically related.
- **E&M** Ear and Mouth. A basic analog signaling method. E&M lead signaling is a specific form of interface between a switching system and a trunk in which the signaling information is transferred across the interface via two-state voltage conditions on two leads, each with a ground return, and separate from the leads used for message information.
- **End Path Delay** Used in this document to mean the time it takes for a signal to pass from the echo canceller to the point of echo and back.
- **ERL** Echo Return Loss. The difference in strength between the original signal and the echo being returned, minus the loss incurred when a signal traverses a hybrid. ERL is measured in dBm.
- **ERLE** The efficiency of the echo canceller, measured in decibels. ERLE is the attenuation added to the ERL.
- **Front-end Clipping** Front-end clipping is defined as the first part of words not being transmitted in the speech stream. Front-end clipping occurs when the first part of syllables (talk spurts) are not recognized by the speech detector.
- **Hybrid** A circuit that converts between the 4-wire and the 2-wire local loop.
- **HNGTM** Hangover Time. The amount of time voice activity detection (VAD) stays on after speech is no longer detected. A longer hangover time will smooth out choppiness but consume more bandwidth. A shorter hangover time will add choppiness and decrease bandwidth consumption. HNGTM only applies to connections using VAD. Hangover time is hardcoded in the UVM at 500 milliseconds.
- **ISDN** Integrated Services Digital Network.
- **LD-CELP** Low Delay Code Excited Linear Prediction.
- **MF** Multifrequency. Push-button telephone signaling that uses two of six possible tones to encode ten digits and five special auxiliary signals.
- **MGCP** Media Gateway Control Protocol as specified in RFC 2705 .
- **Media Gateway** The Media Gateway performs the mapping and translation function between the IP and telephony networks. It is also responsible for support services and network usage. The gateway is defined in ITU H.323 and in IETF drafts.

- **OAM** Operations, Administration, and Maintenance. Special purpose ATM cells perform fault management, continuity checking, and performance measurement functions.
- **Off-hook** Lifting the handset off the cradle closes the switch hook, and current flows through the phone. The circuit is said to have gone off-hook when the central office is informed that a subscriber requires service. Off-hook is the opposite of on-hook. The terms on-hook and off-hook describe the state of the signaling equipment regardless of the type of signaling used.
- **On-hook** Returning the handset to the cradle opens the switch hook, the current ceases to flow. The handset is now on-hook. On-hook is the opposite of off-hook.
- **QSIG** The QSIG protocol provides signaling for private integrated services network exchange (PINX) devices. It is based on International Telephony Union (ITU) Recommendation Q.931 .
- **Sidetone** Sidetone is an intentional byproduct of the hybrid in the phone (for example, sound is transmitted from microphone to receiver). A portion of the speech is allowed to bleed over into the ear piece so that speakers can judge how loudly they are talking. Two speakers may therefore experience quite different sidetone conditions at their respective ends.
- **Signaling** Signaling is the exchange of information regarding the establishment and control of connections. For example, CAS signaling typically uses 2 bits on a T1 or 4 bits on an E1 to indicate on-hook and off-hook status.
- **Talkspurt** The term used when one party on a telephone call is talking. It applies to VAD from the time when speech is first detected to the end of the hangover time.
- **Tandem** Tandem is used in this document to mean a voice connection that undergoes the demultiplex/multiplex, decompress/compress cycle in an intermediate switch before being routed to the destination.
- **VAD** Voice Activity Detection. The process used on CVM or UVM hardware to determine whether a party is talking on one end of a phone. If no party is talking, usually no data is transmitted and a significant bandwidth saving can be achieved.
- **VISM** Voice Interworking Service Module. The Cisco Voice Interworking Service Module (VISM) Release 1.5 is a front and back card set designed to operate on the Cisco MGX 8850 Wide Area Edge Switch platform. The VISM supports VOIP Switching, VOIP Multiservice Access with Call Control, and AAL2 trunking.
- **VOIP** Voice Over Internet Protocol. VOIP is used in this document to mean the transmission of voice traffic in packet form.
- **VNS** Voice Network Switching. A product from Cisco that intelligently routes voice calls across a Cisco WAN Switching network.

Command List

The following commands are referenced in this Tech Note. Commands that have a different syntax for earlier releases of switch software are indicated in parentheses (). Due to functional differences in the cards, the CVM command structure differs from the UVM command structure as follows:

- CVM commands are in a slot.channel or a slot.channel_a–channel_z format (for example, 4.1–24)
- UVM commands are in a slot.line.channel or a slot.line.channel_a–channel_z format (for example, 4.1.1–24)
- VISM commands use a different syntax and are presented in the VISM section.

All commands that require SuperUser-level access are indicated by an asterisk (*). All commands that require Service-level access are indicated by a double asterisk (**).

User Commands	SuperUser-level and Service-level Commands
addcon	cnfcdpparm*
cnfcassw	cnfclnsigparm*

cnfchadv	cnfemb**
cnfchdl	cnfecparm*
cnfchec	cnfnodeparm*
cnfchgn	cnfswfunc**
cnfchutl	cnfuvmpchparm*
cnfchvad	cnfvchparm*
cnfclksrc	dchst*
cnfcond	dspchstats*
cnfln (cnfcln)	dspeparm*
cnflnalm	dspsig*
cnflnpass	off1/on1**
cnfrevsig	
cnfvchtp	
cnfxmtsig	
dspchec	
dspchvad	
dspconst	
dsplncnf	
dsplnerrs (dspclnerrs)	
upln (upcln)	

CVM

This section describes the procedure for tuning voice connections in the IGX 8400 series switch using the CVM. It is assumed that the reader is familiar with the **addcon** command required to create a voice connection in an IGX 8400 network.

There are three models of the CVM card: Model A , Model B, and Model C. The major difference between Model A and Model B is that Model B allows dynamic noise (or pink noise) injection. Noise is played out in the ear piece while the remote side is not talking, which gives the impression that the line is present. With the Model B card, it is possible to dynamically match the noise at the remote end of a telephone conversation without passing large numbers of management packets. The noise level is measured at the remote end and a message is passed back to the source. A representation of the level is then played into the ear piece. This feature is enabled by using the **cnfvchparm** command and setting *Bkgnd Noise* to zero. The CVM Model C card is used for connecting contiguous bundles of up to 24 timeslots across an IGX network. Model C is primarily used for legacy data applications. The Differences in Functionality Between CVM Models on an IGX Switch Tech Note provides more information about CVM model differences.

Configuring an E1 Circuit Line to a PBX or Channel Bank

It is assumed that the PBX configuration details are available. If they are not, some guidelines are provided in the PBX Specifics section of this document.

To configure an E1 circuit line to the PBX using the **cnfln <slot_number>** command, you need to know the physical connection, ground requirements, whether CRC is used, and the type of PBX signaling.

The physical E1 type connection to the PBX can be either BNC or DB15 and can be grounded or not grounded. The BNC connection is 75 Ohm unbalanced and the DB15 connection is 120 Ohm balanced. If the physical connection to the PBX requires that earth ground be disabled, do the following:

- for the E1 BNC interface, remove the nuts from both the Tx and Rx connections of the BNC sockets on the CVM E1 backcard. Then use the **cnfln <slot_number>** command and select the **75 ohm no gnd** option.
- for the E1 DB15 interface, remove the nuts from both the TX and Rx connections of the BNC sockets as there are no other earthing options on the CVM E1 backcard. An incorrectly configured E1 DB15 interface will not function at all.

If the physical connection to the PBX requires earth ground, leave both TX and Rx nuts in place. The G.703 convention is to connect the TX nut to earth ground, but this does not provide any operational benefits on the CVM E1 backcard.

An incorrectly configured E1 BNC interface that requires no earth ground exhibits framing errors in the **dsplnerrs <slot_number>** display after the line has been activated using the **upln <slot_number>** command. A correctly configured E1 BNC interface exhibits no errors in the **dsplnerrs <slot_number>** screen.

The next stage to consider is whether there is error detection on the voice channels or timeslots. Protection of data on the timeslots is performed by running a CRC (called CRC4) in Timeslot 0. An incorrectly configured E1 will result in *CRC Errs* in the **dsplnerrs <slot_number>** display on the IGX 8400 or in CRC errors on the PBX.

The final configuration step is to determine whether the PBXs are using Channel Associated Signaling (CAS) or Common Channel Signaling (CCS) and to reflect the setting on the IGX 8400 using the **cnfln** command.

Following are the main differences between CAS and CCS:

- E1 CAS signaling uses ABCD bits for each channel that are passed constantly in Timeslot 16
- E1 CCS signaling uses a framed protocol passed in Timeslot 16 that sends indications such as *off-hook* only when a change occurs
- E1 CCS signaling is feature rich; for example, with Q.931 and DPNSS there are many supplementary services such as *camp on*.

CAS

If CAS is selected, the IGX automatically routes the ABCD signaling bits between the PBXs on the connection. No **addcon** command for timeslot 16 is allowed. This works for point-to-point configurations and the more complicated point-to-multipoint networks. Set the **cnfvchtp <channel_number>** command to monitor timeslot usage. When **cnfvchtp** is correctly configured to match the PBX signaling, the **dsponst** screen indicates the status of a connection (for example, on-hook, off-hook, modem upgrade). To determine the PBX signaling, issue the **dspsig <channel_number>** command to view snapshots of the signaling states, which can then be configured in **cnfvchtp**.

If the PBX is pulse dialing on the signaling channel, the **cnfchdl** command is used to configure out-of-band signaling to pass the signaling without distortion. When there are dissimilar signaling systems between the PBXs, use the **dspsig** command to obtain the signaling states and the **cnfrevsig** and **cnfxmtsiz** command to manipulate the signaling bits. For example, to convert from T1 E&M to E1 SSDC5a signaling the following settings may be used:

- **cnfxmtsiz** <slot_channel> **I I T I** (at the T1 E&M end)
- **cnfxmtsiz** <slot_channel> **I 1 0 1** (at the E1 SSDC5a end)

The conditioning criteria to apply to the signaling bits when the connection is derouted can be configured using the **cnfcond** command. Configuring **cnfcond** allows a defined pattern to appear on the ABCD signaling bits when the connection fails. This command also allows timed pulses to be applied to the signaling to ensure the PBX returns to a known state.

CCS

If CCS is used, a transparent connection must be added between two PBXs even though the data is framed. The CAS multipoint feature is not supported with CCS on the CVM. CCS is enabled using the command **addcon** <slot.16 node slot.16 t> where *slot* refers to the CVM card position and *node* refers to the remote IGX 8400. In addition to the **addcon** command, ensure that **cnfvchtp** <slot.16> is set to *No Sig* at each end of the connection in the IGX 8400 network. If **cnfln** is incorrectly configured as CAS, a CCS PBX will not work. A CAS PBX will work if **cnfln** is incorrectly configured for CCS, but bandwidth will be wasted because the ABCD bits will pass through continuously.

CCS circuits do not allow for individual timeslot or signaling state monitoring using the **dsponst** or **dspsig** commands on the IGX 8400.

Clocking Considerations

The normal clocking condition to the PBX is *normal*, which implies that the CVM times the TX data and expects the Rx data frequency to match. This means that the CVM is providing clock to the PBX and that the PBX is using the receive timing to clock transmit data out to the CVM. To configure, set **cnfln** to *Loop clock: No* on the IGX 8400 and PBX to *loop clock*. If the PBX is connected to a digital ISDN service or a Building Integrated Timing Supply (BITS), then it is acquiring a clock reference from another source. In this case declare the PBX to be a clock source to IGX using the **cnfclksrc** command. If the PBX is not connected to ISDN, BITS, or another known clock source, do not declare it as a clock source. To ensure that the PBX clocking is consistent with the configuration:

1. Refer to the **dsplnerrs** screen to ensure the clocking is not causing frame slips. The **cnfln** command may be required to adjust the clock configuration to *Loop* or *Local*.
2. Verify that the PBX is not detecting frame slips.
3. Use the **cnflnaln** command to make the alarming of both circuit line and trunk alarms more sensitive so the operator is made aware of any problems.

Enabling the Circuit Line

After the physical and protocol sides of the circuit line have been configured, bring the circuit online using the **upln** command. After a few seconds the **dsplns** display should show *Clear – OK*. If there are minor or major alarms, check the physical interface and the **cnfln** parameters.

Use the **dsplnerrs** command to determine if the link is working correctly. The information provided by the **dsplnerrs** command is summarized below.

Statistical Alarms	Integrated ("hard") Alarms
Bipolar Errors – number of times two consecutive pulses have the same polarity (T1 lines only).	Loss of Signal (RED) – signal level at receive input is below threshold.

Frame Slips – number of times a frame is inserted or deleted to reestablish synchronization. This is commonly caused by a clock mismatch between the PBX and the IGX 8400.	AIS (BLU) – a string of 2048 or more consecutive ones has been detected. This is known as a 'keepalive' signal sent in the downstream direction of a fault.
Out of Frames – number of times a loss of frame synchronization is detected on this circuit line.	Out of Frame (RED) – frame synchronization loss.
Loss of Signal – number of times that the signal level at the circuit line input went below the minimum acceptable level.	Remote Out of Frame (YEL) – far end receiver out of frame.
Frame bit errors – number of times the frame bit failed to alternate (E1 lines only).	
CRC errors – number of times the generated CRC character did not match the received CRC character. (CRC checking must be enabled on E1 lines using the cnfln command.)	
AIS-16 – number of times the Alarm Information Signal (Blue alarm) was received (E1 lines only).	
Out of Mframes – number of times a multiframe synchronization error was detected (E1 lines only).	

Modem Support

Modem traffic is different from voice traffic in that voice traffic consists of peaks and troughs of volume and has a mathematically modeled variance. The CVM voice compression algorithms do not work well with high speed modems. To avoid negative impact to modem traffic, after the CVM detects a modem (for V.25 modems this is typically a 2100 Hertz tone), the connection is upgraded from the current configuration to a pulse code modulation (PCM) clear channel for the duration of the modem call. If voice is detected on the next call, the connection is then downgraded to the original configuration (for example, c32) to re-enable bandwidth savings.

The IGX 8400 routinely polls all the CVMs and UVMs to monitor the status of modem calls. The modem polling interval can be adjusted using the **cnfnodparm** command or modem polling can be disabled using the **off1** command.

For tuning modem calls, the **cnfdpparm** command must be configured at both ends of the connection to:

- adjust the modem/fax detection silence maximum (*MDM Detect Silence Max.*) from 0C Hex (one second) to 24 Hex (three seconds). The modem/fax detection silence parameter defines the amount of

time a channel stays in a modem/fax detected state.

- adjust the modem stationary coefficient (*MDM Stationary Coef.*) from 14 Hex to 25 Hex. The modem stationary coefficient is used to differentiate between slow modems (< 4800 baud) and fast modems (>4800 baud).

The bandwidth of the connection upgrade also affects modem performance. The **cnfvchparm** command must be configured at both ends of the connection to enable a compressed voice connection (for example, c32) to be upgraded to:

1. A 64 kbps clear channel connection that will work with any type of fax or modem.
2. A 32 kbps ADPCM connection that is optimized for faxes at 9600 bps.

Adjusting Gain / Loss on CAS Connections

To troubleshoot CAS connections using test calls, it is necessary to identify the timeslot the PBX is using. PBXs typically seize 64-kbps trunks randomly and do not select the same trunk for multiple calls. This dynamic behavior can prolong testing. Some PBXs can be configured to seize only one trunk during out-of-service testing, but if the PBX technician or a maintenance window is not available, the following procedure can be used to troubleshoot connections.

1. Dial a fax machine at the far end that has a handset and at the same time watch the **dsponst** screen.

When the remote fax answers, an "M" appears in the timeslot the PBX has selected. After the handset is taken off-hook from the remote fax, the CVM downgrades the call and the M eventually goes away. After the M is gone from the **dsponst** screen, a voice call has been established.

2. Continuously press the "#" key on the local telephone keypad. If the telephone does not generate a continuous tone, find one that does and start again.
3. Use the **dchst <slot.channel> <1>** command to display the power level received from the handset. The receive level should be -13 dBm.
4. Calculate the number of dBs of gain/loss needed to have a receive level of -13 dBm. If the level received is not within +/- 3dB of this figure, adjust the output levels of the PBX so that the received level is -13 dBm.
5. If PBX gain adjustment is not possible, use the **cnfchgn <slot.channel>** command to adjust the receive level to -13dB by inserting loss or gain into the input of the CVM. Use the **dchst <slot.channel> <1>** command to confirm that the level is correct.

At the far end of the connection, use the same commands to configure gain/loss in the CVM output to compensate for the loss/gain inserted at the input. This is to ensure that the signal leaves the IGX 8400 at the same level it came in. The IGX 8400 network must have a flat response and insert no loss or gain.

6. Ensure the signal level is similar when calls are made from a variety of handsets, or from *offnet* or remote locations. If the levels vary widely, review the voice loss plan.
7. Repeat this process in the opposite direction.

Do not assume that the gain/loss values will be identical for both directions. Many PBXs are configured differently even in the same network.

8. After the gain/loss is configured for transmit/receive paths at each end of the voice connection, configure the remaining connections with the same settings.

Voice Activity Detection

Voice Activity Detection (VAD) is the most complicated algorithm implemented on the CVM. The VAD function requires the CVM to constantly monitor every voice channel to detect the presence of voice or

modem activity. Depending on the type of connection configured for the channel, VAD determines whether to build and transmit fast packets for the connection. The suppression of fast packet transmission in the IGX 8400 network results in bandwidth savings.

The following are VAD connection types:

- v
- c32
- c24
- c16
- c16z

The VAD algorithm performs the opposite function of the Adaptive Voice algorithm. Adaptive voice was useful for the CVM Model A with VAD connections that originated in a combination of quiet and noisy rooms. In that environment, the CVM VAD static background noise injection algorithm was suboptimal. However, the CVM Model B uses a dynamic background noise matching algorithm that greatly improves VAD performance. For all CVM Model B connections using VAD, Adaptive Voice must be disabled using one of the following commands:

- **cnfswfunc** (per node)
- **cnfchadv** (per connection)

Configuring Channel Utilization

When a VAD connection is added, the default channel utilization is 60 percent. Percent utilization is used as a factor in building a *load model*. Each IGX 8400 maintains a static load model of connection bandwidth and resource requirements. Based on the static load model, decisions regarding connection routing are made. If a trunk does not have the bandwidth available to support a target connection, an alternate trunk must be found or the connection will de-route and traffic will stop. If a network has a lot of callers in noisy conditions, utilization on voice connections may exceed 60 percent. In this case, the percent utilization must be increased to reflect actual use. If the load model does not reflect actual use, there may be voice (VAD connections) or non-timestamped (non-VAD connections) packet drops on network trunks resulting in poor voice quality. The command **cnfchutl** is used to increase or decrease channel utilization.

Fine Tuning VAD

The default VAD settings work well for most connections. For environments that require additional connection tuning, follow these steps:

1. Locate a point with average background noise characteristics in the location under test.
2. From the test point in the building with *average* listening conditions, place a call to a similar location. Identify the timeslot the PBX is using for the call. If necessary, follow the steps listed in the Adjusting Gain / Loss on CAS Connections section to determine the timeslot.
3. Issue the **dchst <slot_number.connection_number> 1** command and watch Registers 1 and 2 to determine whether cells are generated when the two parties speak. If either party stops speaking (for example, leaves the handset close to their head without putting their hand over the mouthpiece), confirm that cell generation stops.
4. If cells do not stop when the speaker is silent, adjust the VAD threshold with the **cnfchvad** command. Adjust the *VAD Mid Power* and *VAD Low Power* parameters to raise the VAD threshold. Note that the lower the VAD threshold, the more FastPackets generated, and the more trunk bandwidth that's required. The increase in required bandwidth must be reflected by increasing connection utilization using the **cnfchutl** command.
5. If the VAD threshold is too high, front-end clipping will be experienced. Make test calls from different points in the building to ensure that VAD operates satisfactorily. The efficiency gained by

VAD varies among calls and among handsets. It is most important to ensure that the average call has good efficiency and that all callers experience good quality.

Details of the **dchst <slot_number.connection_number> 1** screen are shown below. Note that Channelized Data Pad (CDP) and Channelized Voice Module (CVM) are used interchangeably.

```
i3          TRM   SuperUser      IGX 8420  9.1.13   Mar. 21 2000 20:05 CST
```

```
Channelized Data Pad state display for channel 16.1          Snapshot
```

```
Transmit dBm0: -70.0          Level of signal transmitted to the CLN
```

```
Receive dBm0: -67.0          Level of signal received from the CLN
```

```
Register 0 = 2B2D          TX PCM Value (MSB) | RX PCM Value (LS byte)
```

```
Register 1 = FFFF          TX Packet count (# of packets transmitted to Cell Bus)
```

```
Register 2 = FFFF          RX Packet count (# of packets received from Cell Bus)
```

```
Register 3 = 1583          DSP # to which the current connection is assigned
```

```
Register 4 = 0000          Lost packet count for G.729 (g729r8) and G.728 (116) connections
```

```
Register 5 = 3601
```

```
Register 6 = 160C
```

```
Last Command: dchst 16.1 1
```

The default VAD settings are provided below. CDP and CVM are again used interchangeably.

```
i3          TRM   SuperUser      IGX 8420  9.1.13   Mar. 21 2000 19:30 CST
```

```
CDP Models All
```

VAD												V.25
Sample Bkgnd				Power Thresholds			ZCR		Stat. Hang Pri			Detect
>From	Delay	Noise	HPF	High	Mid	Low	High	Low	Coef.	over	Float	upgrade
16.1-9	A8	67	ON	3160	40	40	50	15	30	42	ON	64K
16.12-24	A8	67	ON	3160	40	40	50	15	30	42	ON	64K

The following table lists the hexadecimal values for integer dBm0 values used for the following parameters:

- VAD High Pwr Thrsh (**cnfchvad**)
- VAD Mid Pwr Thrsh (**cnfchvad**)
- VAD Low Pwr Thrsh (**cnfchvad**)
- MDM Low Pwr Thrsh (**cnfcdpparm**)

dBm0	Hex Value	dBm0	Hex Value	dBm0	Hex Value
-24	F956	-41	04E9	-58	0019
-25	C60E	-42	03F3	-59	0014
-26	9D52	-43	0323	-60	0010
-27	7CF7	-44	027E	-61	000C
-28	6343	-45	01FB	-62	000A
-29	4ED9	-46	0192	-63	0008
-30	3EA1	-47	013F	-64	0006
-31	31BF	-48	00FE	-65	0005
-32	2784	-49	00C9	-66	0004
-33	1F63	-50	00A0	-67	0003
-34	18EF	-51	007F	-68	0002
-35	13CE	-52	0065	-69	0002
-36	0FBB	-53	0050	-70	0001
-37	0C7F	-54	003F	-71	0001
-38	09ED	-55	0032	-72	0001
-39	07E2	-56	0028	-73	0000
-40	0643	-57	001F		

Video Teleconferencing on a CVM With VAD

The CVM does not directly support video teleconferencing. The CVM can provide bandwidth savings for video teleconferencing connections routed over the CVM to PBX connection. The steps to configure a connection to support video teleconferencing are as follows:

1. Connect the number of channels needed to support video traffic and configure the PBX to bar these channels for voice.
2. Disable the Echo Canceller on the connection using the **cnfchec** command.
3. Add connections in the IGX network as ?v? type using the **addcon** command.
4. Set the gain to zero using the **cnfchgn** command.
5. Set the delay to Hex 01 and disable the high pass filter using the **cnfvchparm** command.

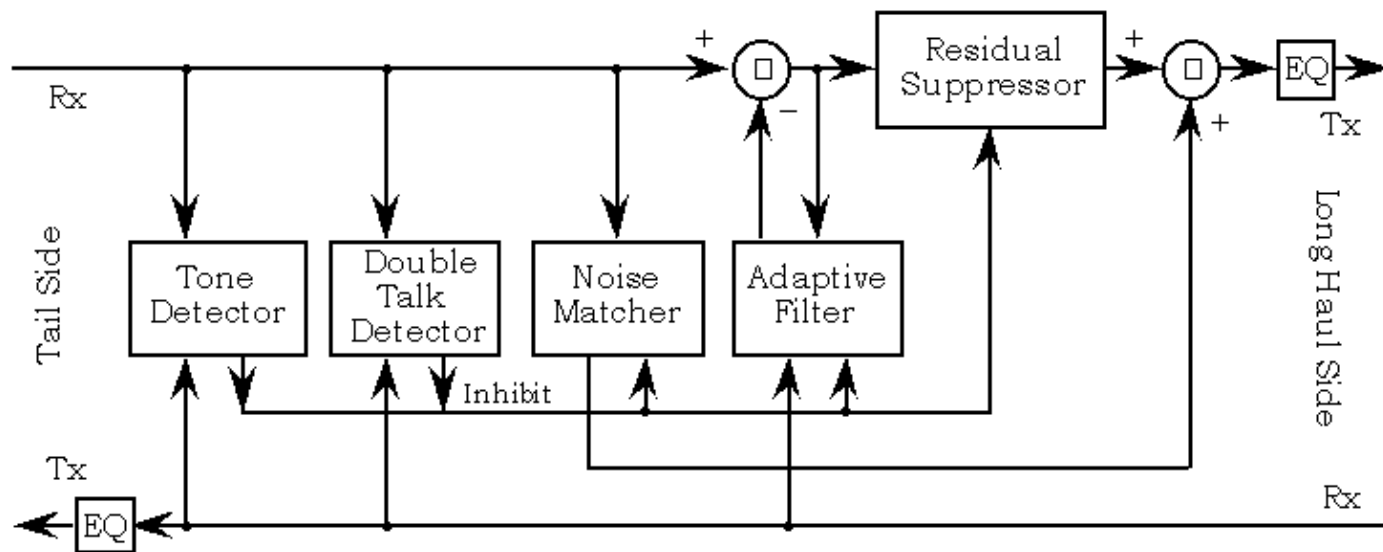
When no video is being transmitted from the PBX, VAD detects silence and suppresses fast packet generation.

Echo Canceling

Echo cancellers are used to eliminate echo caused by 2-wire to 4-wire converters or hybrids in a telecommunications network. An echo canceller achieves this by:

- Modeling the measured echo on individual voice channels.
- Subtracting the measured echo (echo replica) from the reflected signal.
- Continuously adapting to the echo (convergence).
- Recognizing the difference between echo and speech.
- Disabling echo cancellation when modems are used.

The following diagram illustrates how an echo canceller functions. Note that the algorithm is independently performed on each channel (DS-0) of a T1 or E1 signal. Thus, the echo that was introduced in the analog portion of the circuit is eliminated in the digital portion of the circuit.



The echo canceller is inserted between the circuit line termination of the IGX and the connected PBX or channel bank. The echo canceller continuously observes the signal (speech) going from the IGX to the PBX (transmit direction). The echo canceller stores the transmitted signal and compares it to the received signal. Choosing moments when there is no speech in the receive direction, the echo canceller assumes that all the energy coming from that direction is echo caused by reflections at the 2-wire termination on the tail side of the call. Therefore, the signal should be a delayed, attenuated version of the original signal that is already stored. The echo canceller uses the DSP to calculate the delay and reduction in the original signal necessary to completely cancel out the received signal. This process is called *convergence* and is used to create a mathematical model of the echo delay and amplitude of the echo in the tail circuit. The calculation is then applied continuously to the call, reducing the reflected portion of the received signal by at least 30 dBm.

The echo canceller at each end of a call reduces the echo in each tail circuit so that the echo is imperceptible, even at the level of delay introduced by the IGX 8400. If PBX hybrid balance is good, set the echo return loss to low using the **cnfchec** and **cnfecparm** commands to improve the convergence time of the echo canceller. It is important for the signal level coming into the IGX to be set correctly to get the best voice quality, best efficiency for the VAD algorithm, and best performance for the echo cancellers. To set the correct gain/loss in the IGX, you must make a test call and measure the signal strength level using the **dchst** command.

Echo Canceller Limitations

Due to the differences in paths and terminations, the convergence process must be repeated at the beginning of each call. The echo canceller uses signaling information and speech energy to determine when a call is

beginning. While it is possible to configure some cancellers to converge based on changes of the signaling bits, most cancellers attempt to converge continuously whenever speech is present. When coupled with VAD, the echo canceller will attempt to converge at the beginning of each talkspurt. In conditions of high reflected signal (low echo return loss), this can result in the talker hearing echo at the beginning of phrases.

For call paths with echo return loss (ERL) higher than 6 dBm, set configurable echo cancellers to a value of 0. For call paths with low ERL (6–10 dBm), use the value of 6. The canceller can converge much faster if the ERL is known. If the ERL diverges from the configured value, the canceller will have great difficulty converging and bad echo will result. Convergence may take from 20 to 200 milliseconds.

Another difficult situation for the echo canceller is double-talk. It is impossible to run the echo calculation when both callers are talking. Therefore, the echo canceller must recognize double-talk and continue cancellation based on the information before double-talk was detected. Echo cancellation may be poor or other anomalies may occur if double-talk is detected too late or not at all.

Echo Canceller Options

There is usually some form of residual suppression, center-clipper, or nonlinear processing feature in echo cancellers. This feature recognizes that signals with very low power are usually mixed up with noise. To guard against some of this noise being echo, the canceller suppresses it all and transmits idle code instead. This may give rise to clipping on quiet calls, particularly when double-talk is present and the two directions of the call have very different power levels.

An enhancement that center-clipper provides is noise matching. The noise matching function recognizes that some calls may suffer from choppy voice due to the level of background noise during speech being changed to silence while the signal is clipped. The noise matching function continuously samples the noise level before the echo cancellation in the receive direction, and injects an appropriate level of noise after the clipper. The listener no longer hears noise discontinuities due to the center-clipper. Noise matching should generally be left enabled in the echo canceller, even on VAD connections. This function prevents the remote listener from hearing background noise discontinuities caused by the IGX building and transmitting packets during silent periods, such as during VAD hangover time (HNGTM).

Echo cancellers have a tone detection feature to identify fax and fast modem calls. Echo canceling is disabled when a 2100 Hz tone is detected and is not re-enabled until the end of the call. The end of the call is identified by signal power decreasing below a threshold. For normal applications, this feature should be enabled.

CVM Integrated Echo Canceller

The CVM supports an optional 24-channel or 32-channel integrated echo canceller (IEC) which provides:

- Echo Return Loss Enhancement (ERLE) > 30 dB
- Convergence time of < 50 ms
- Tone disabler
- Center clipper
- Matching Noise Injection

on tail circuits with:

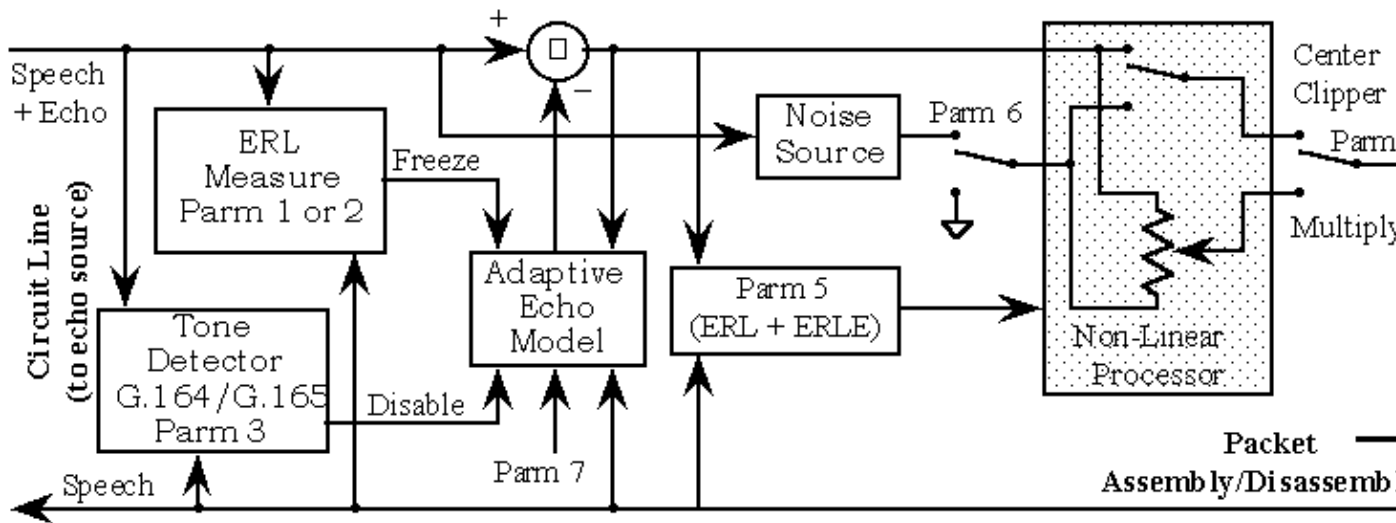
- Echo Path Delay up to 32 milliseconds
- Echo Return Loss (ERL) > 0 dB

The IEC performs the same function as an external echo canceller. However, the integrated echo canceller is located between the gain/loss insertion circuitry and the packet assembly and disassembly circuitry. Note that

the command **dchst <slot_number.channel_number>** displays the input and output dBm levels at the point between the gain insertion and the echo canceller.



The IEC provides basically the same configurable internal options as a Tellabs echo canceller. The **dspecparm** and **cnfecparm** commands are used to monitor and configure the parameters of the optional IEC on the CVM. The configurable options are referenced to the corresponding parameters of the **cnfecparm** command described below.



i3 TRM SuperUser IGX 8420 9.1.13 Mar. 22 2000 16:41 CST

IEC Slot 16 Parameters

```

1 IEC Echo Return Loss High (.1 dBs) [          60] (D)
2 IEC Echo Return Loss Low  (.1 dBs) [          30] (D)
3 IEC Tone Disabler Type      [          G.164]
4 IEC Nonlinear Processing    [Center Clipper]
5 IEC Nonlinear Processing Threshold [          18] (D)
6 IEC Noise Injection         [          Enabled]
7 IEC Voice Template          [          USA]
    
```

Last Command: cnfecparm 16

Parameters 1 and 2 specify options which can be selected for each channel of the circuit line in multiples of 0.1 dB. The echo return loss value selected represents the minimum ERL required for the echo cancellation circuit to be enabled. If the measured ERL is less than the value specified, the signal is not considered echo and the echo convergence mechanism is frozen, even though echo is still canceled based on the most recent echo model.

Parameter 3 allows selection of the G.164 or G.165 tone disabling protocol to support high-speed modem transmission. The G.164 protocol is the older mechanism originally developed for echo suppresser technology. It requires the detection of a 2100 Hz sine wave to disable the echo canceller. The G.165 protocol requires the detection of a 2100 Hz sine wave with phase reversals every 450 milliseconds. Two of these phase reversals are required to disable the echo canceller. Low speed modems generate the 2100 Hz sine wave and high speed modems generate the 2100 Hz sine wave with phase reversals. This implies that G.164 can work with both low and high speed modems. It is recommended to use G.165 for most connections.

Parameter 4 allows selection of either the standard center clipper technique or the newer multiplying technique. In the conventional center clipper mechanism, any post-cancelled signal below the threshold specified in Parameter 5 is eliminated using an abrupt transition. If it's enabled in Parameter 6, it's replaced by low-level synthesized noise. Using the multiplying technique, the transition from signal to injected noise is done slowly, over a period of approximately one second. It is recommended to use the center clipper technique for most connections.

Parameter 5 specifies, in dBm, the threshold at which the nonlinear processor is activated. If the ERL of the tail circuit plus the ERLE provided by the echo canceller exceeds this value, the remaining signal will be eliminated.

Parameter 6 allows the user to enable or disable the noise injection function. If this function is disabled, silence is sent when the nonlinear processor is activated. When nonlinear processing is enabled in the center clipper mode, noise injection is optionally enabled to fill the periods when the echo signal is being cut off. The level of the noise injected is dynamic. It is approximately equal to the background noise content of the current connection.

In a VAD application noise injection should be disabled since the IEC inserts its noise from the far end of the network. Packets would have to flow to get the artificial noise to the listener. If you are going to use dynamic noise insertion, disable the echo cancellers injected noise. In addition, using the **cnfvchparm** command set the inserted noise level to "0." This enables the dynamic noise insertion feature of the CVM Model B card. (If you have model A cards, please contact your supplier. These cards will not work with dynamic noise injection). Reset the card with the command **resetcd <slot_number> h** to ensure the parameters are downloaded to the card. The noise injection schemes used by the Model B card and the echo canceller are different.

Parameter 7 allows the user to select the USA or UK echo cancellation template. The UK template is provided solely for better performance in environments using analog tail circuits, which are typical in the UK. This provides a high-power input into the network. The USA setting should be interpreted as a low power input.

The **dspchec** and **cnfchec** commands allow the user to monitor and specify the parameters that determine the operations of a single channel or a range of channels of the optional IEC on the CVM. The **cnfchec** command allows per-channel configuration to:

- Enable or disable echo cancellation.
- Select the high or low minimum ERL set with the **cnfecparm** command.
- Enable or disable the disabling of the echo cancellation due to tone detection.
- Enable or disable the convergence function.
- Enable or disable the nonlinear processor function.
- Display the voice template selected for the line with the **cnfecparm** command.

Channels	Echo Cancel	Echo Return Loss(.1 dBs)	Tone Disabler	Conver- gence	Nonlinear Processing	Voice Bkgrnd Tplmt	Filter
16.1	Enabled	Low 30	Enabled	Enabled	Enabled	USA	-
16.2-24	Disabled	High 60	Enabled	Enabled	Enabled	USA	-

Last Command: cnfchec 16.1

- Echo Cancel [enable|disable]. Enable or disable IEC. The IEC is bypassed by "disable".
- Echo Return Loss [high|low]. Selects one of two options configured in the **cnfecparm** command. The IEC will not converge while the echo signal is within this amount of the speech signal. If this setting is higher than the ERL, the IEC will not converge. Choose "high" only if the ERL is at least 3 db better than the "high" setting configured using the **cnfecparm** command.
- Tone Disabler [enable|disable]. The tone disabler feature allows the IEC to detect a preamble tone associated with dial-up modems and disable itself when a modem is detected. This is essential for high speed full-duplex modems.
- Convergence [enable|disable]. Disabling the convergence function of a channel has the effect of freezing the echo canceller in its current state, preventing it from doing any further improvement or modification of the normally adaptive echo cancellation process. This configuration is typically used only for troubleshooting.
- Nonlinear Processing [enable|disable] Since there is always a small amount of echo that bypasses an echo canceller, it is sometimes desirable to process this residual echo in a nonlinear fashion. If nonlinear processing is enabled, the IEC stops sending all data when the echo signal is sufficiently below the speech signal. There is a configurable threshold that defines how far below the speech signal the echo signal must be before the nonlinear processing begins.
- Voice Template [USA|UK]. Each of these template choices represents a set of internal IEC parameters which are not otherwise available to the user. The USA template is optimized for voice levels approximately from -10 dBm0 to -50 dBm0. The UK template is optimized for voice levels that reach above -10 dBm0 to +3 dBm0. When the UK template is chosen, IEC performance on voice levels in the -10 dBm0 to -50 dBm0 is compromised. The UK template should only be used when voice levels are extremely high.

Echo Reduction Measures

Echo cancellers represent a relatively complex solution to a complex problem. However, there are some straightforward measures which can improve their effectiveness.

1. Verify that all echo cancellation parameters are correct.
2. Reduce, as much as possible, the level of echo (ERL) seen by the echo canceller. Adding loss in the tail circuit is always helpful. Sometimes it's possible to find the specific 2-wire termination causing echo. Echo can be improved by line build-out or impedance options on trunk cards. Replace 2-wire circuits with 4-wire circuits to eliminate hybrid echo.
3. Echo cancellers can usually accommodate up to 32 milliseconds of tail circuit delay. If the delay is close to this limit, an extended version of the echo canceller may be needed.
4. Echo cancellers have difficulty with double-talk when the signal levels in each direction of a call differ by more than 10 dB. It may be possible to change the network loss plan to allow for this signal level.

5. Delay introduced by the IGX 8400 may be reduced on VAD connections by configuring Sample Input Delay from A8 to 50 if only on-net calls are made. Reducing delay may improve echo canceller performance.
6. IGX 8400 network changes to route voice connections over the smallest number of hops and to balance load evenly across trunks will reduce delay and may improve echo canceller performance.
7. For troubleshooting purposes, test the problem connection with VAD disabled and again as an uncompressed connection (p-type) to isolate the source of echo.

UVM

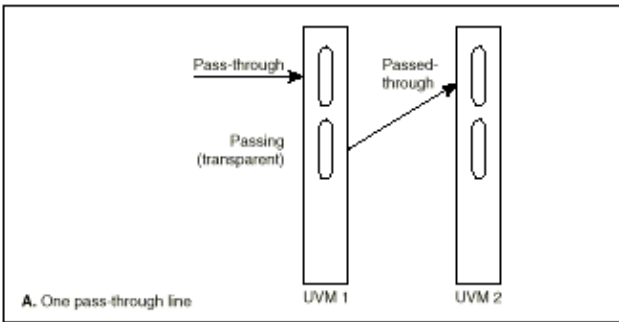
This section describes the procedure for tuning voice connections in the IGX 8400 series switch using the UVM. It is assumed that the reader is familiar with the **addcon** command required to create a voice connection in an IGX 8400 network.

The UVM was introduced in switch software release 8.2.5x and is also supported in releases 8.5, 9.1, 9.2, and later. The feature set of the UVM varies depending on the release of the switch software. The features and performance discussed in this section are with respect to switch software release 9.1.13 and later using UVM firmware Model E version D (DED).

The UVM is the next generation voice card for the IGX. Most of the commands used for the CVM apply to the UVM. For example, the **dchst <slot_number.line_number.channel_number>** command is used to monitor signal power levels. The sections relating to line features for the CVM are the same and are not repeated here. UVM features include:

- Channelized T1/E1/J1 interfaces
- Voice coding types
 - ◆ PCM
 - ◆ ADPCM (G.726): 32 channels per UVM
 - ◆ LD-CELP (G.728): 16 channels per UVM
 - ◆ CS-ACELP (G.729): 16 channels per UVM
 - ◆ CS-ACELP (G.729A): 32 channels per UVM
- Two line interface ports
- Voice activity detection (new per channel **cnfvmchparm** command)
- Integrated echo cancellation on back card using Mitel chipset.
- Fax relay support for G.729 connections (new **cnfchfax** command)
- Modem detection
- Super-rate data connections
- Voice switching
- CAS switching only for VNS support (refer to the VNS section of this document)
- D-channel compression for CCS support using UVM Firmware DED and later
- D-channel compression for VNS signaling support (refer to the VNS section of this document)
- Y-cable redundancy
- CAS or CCS signaling
- Pass through (new **cnflnpass** command)

Pass Through



The UVM supports 16 channels configured for G729 voice compression. To allow for full T1 or E1 capability, two UVM card sets must be chained together. To chain UVM cards, add an external cable between two UVMs in the same chassis and link them using the **cnflnpass** command. See the "UVM Pass Through" diagram above.

To configure pass-through, separate channel numbers must be used to denote the UVMs in separate slots. For example, to connect to a PBX with an E1 CCS interface using UVMs in slots 12 and 13, issue the following commands:

1. **upln** 12.1 the Passing line
2. **upln** 12.2 the Blocking line
3. **upln** 13.1 the Inserting line
4. **cnflnpass** 12.1 13.1 pass-through requires the primary card to use line 1
5. Repeat commands 1 through 4 at the remote IGX 8400.
6. **addcon** 12.1.1–15 <remote_nodename> 12.1.1–15 g729r8
7. **addcon** 12.1.16 <remote_nodename> 12.1.16 t
8. **addcon** 13.1.17–31 <remote_nodename> 13.1.17–31 g729r8

Within the IGX network there are connections between cards 12 and 13, but on the PBX side there are 30 connected timeslots with signaling to card 12.

Modem Support

The UVM supports fast modem detection and introduces a new feature, fax relay.

Fast Modem Detection

UVM fast modem detection is available in all supported releases of switch software. The UVM V.25 modem detector recognizes the steady 2100 Hz tone output by V.25 fast modems (> 4800 baud) and fax machines to disable echo cancellers at the beginning of transmission. This function is called the fast modem capability and is the default configuration on UVM connections. The monitoring of connections to detect fast modem calls is performed by the switch software *modem polling* function. Modem polling can be disabled using the **off1** command or the frequency of polls changed using the **cnfnodetparm** command. After a fast modem has been detected, the connection can be upgraded from the current compression level to 32 kbps or 64 kbps using the **cnfvchparm** command. Modem polling is used to determine when a modem call has been disconnected so that the connection can be downgraded to the original bandwidth (usually less than 32 kbps or 64 kbps), saving network resources.

To verify that a connection is using fast modem detection, (1) issue the **cnfchfax** command and verify that the Fax field is disabled, and (2) issue the **cnfvchparm** command and verify that the *V.25 Detect* field is not disabled (either 64KB or 32KB). Note that the *V.25 Detect* field is the only field applicable to the UVM for the **cnfvchparm** command.

UVM fast modem upgrade to 32 kbps is not supported for g729ar8 and g729ar8v connections. For g729ar8 and g729ar8v connections, configure the V.25 Detect field to 64KB.

For better fast modem performance, it may be necessary to change the silence duration from 1600 to 5100 or higher. Issue the **cnfuvnmchparm** command and configure the SIL DUR parameter.

For networks with heavy fax use, the *Line pct fast modem* parameter in the **cnfln** command may need to be increased. This parameter is known as the percent fast modem function and it is used by switch software to ensure that there is enough CellBus bandwidth available to the UVM for the large number of FastPackets that are generated for all the simultaneous fax calls. The *Line pct fast modem* default is 20 percent, but this may need to be increased to 40 percent or higher to more closely represent fax usage over the UVM line. If the *Line pct fast modem* parameter is too low, FastPackets will be dropped. This will impact all voice connections on the card (that is, bad voice quality) and cause faxes to be detected but unable to maintain communication as observed using the **dsponst** screen and noting a rapid change from *M* to *+*. The *Line pct fast modem* parameter does not affect the *load model*.

Fax Relay

Fax relay is a new feature. When a fax call is detected, the current voice compression is disabled and replaced by a fax demodulation/modulation algorithm. The algorithm manages fax negotiation and then transports the data across the IGX 8400 network at 9.6 kbps or lower, as the fax bit stream requires. The new **cnfchfax** command can be used to enable or disable the feature on both sides of the connection.

Fax relay can be configured for G.729A connections but it is not supported. This is due to the way that DSPs are allocated on the UVM card. Each DSP will support two G.729A connections but only one fax relay call. If fax relay is needed, use g729 or l16 which uses a whole DSP. This can be confusing as it is possible to configure fax relay on a G.729A, however the firmware will prevent the connection from upgrading.

Fax relay is delay sensitive and connections with long round trip delays may not be able to support it. Long round trip delays are caused by:

- satellite hops
- voice over ATM trunks using a customized **cnfcmf** parameter that increases the wait time for FastPackets

VAD

The operation of UVM VAD is similar to VAD on the CVM. The **cnfuvnmchparm** command is used to tune VAD. Following are VAD connection types:

- v
- c32
- c24
- l16v
- g729r8v (may exhibit poor voice quality over tandem switched networks)
- g729ar8v (may exhibit poor voice quality over tandem switched networks)

```
i2          TRM   SuperUser      IGX 8420  9.1.13   Mar. 23 2000 14:12 PST
```

```
From          Parameter:
              NSE   PIU   VAD   MDM   SIL   SIL
```

13.1.1.1	INJ	LVL	THLD	THLD	DUR	THLD	7	8	9	10	11
13.1.1.1-24	60	0	40	40	1600	40	0	0	0	0	0
13.2.1-24	60	0	40	40	1600	40	0	0	0	0	0

Last Command: `cnfuvvmchparm 13.1.1`

The following default settings for VAD on the UVM displayed in the **cnfuvvmchparm** command may need to be tuned for better voice performance. Changes to the **cnfuvvmchparm** settings must be made at both ends of a connection. Improving voice quality typically comes at the cost of bandwidth savings.

- *NSE INJ* (noise injection). Units are -10 dBm0. Range is 1–15. Typically configured to 8 to represent -80 dBm.
- *VAD THLD* (VAD threshold). Units are -1 dBm0. Range is 1–255. Typically reduced from 40 (-40 dBm) to 45 or lower. Do not configure noise injection and VAD threshold to be the same value.
- *MDM THLD* (modem threshold). Units are -1 dBm0. Range is 1–255. Below this threshold, the modem tone is ignored or not detected. Typically configured at 40.
- *SIL DUR* (silence detection window size). Units are 20 milliseconds. Range is 1–255 (20 milliseconds – 5.1 seconds). Silence is detected if the signal level stays below the silence detection threshold in dBm0 for the duration specified by the silence detection window size in milliseconds. Silence detection is used to determine when a fax/data modem call has terminated.
- *SIL THLD* (silence detection threshold). Units are -1 dBm0. Range is 20–80.
- 7 (enable DC offset filter). For a PBX that sends a non-standard idle code, a DC filter has been added during the computation of bi-directional silence. This silence detection is used for downgrading a connection in the V.25 modem state. The default value is 0 (disabled). To enable the filter, use 1.
- 8 (upper convergence speed threshold.) Used to tune echo convergence threshold. Range is 12 (fastest convergence time) to 30 (slowest convergence time). Note that the fastest setting will increase the instance of reconverge during normal pauses in speech such as the beginnings of distinct sentences. Reconverge during speech may result in a brief period of echo in the middle of a conversation.
- 9 (double talk detection threshold). 0 is equal to the default value of 5 dB. Only configured to improve echo cancellation on circuits with very poor ERL (<5 dB).

The *PIU LVL* setting is now hard-coded in the UVM firmware. The PCM Interface Unit (PIU) value entered by the user must be zero. The default value for this parameter is zero. The value is always displayed as zero.

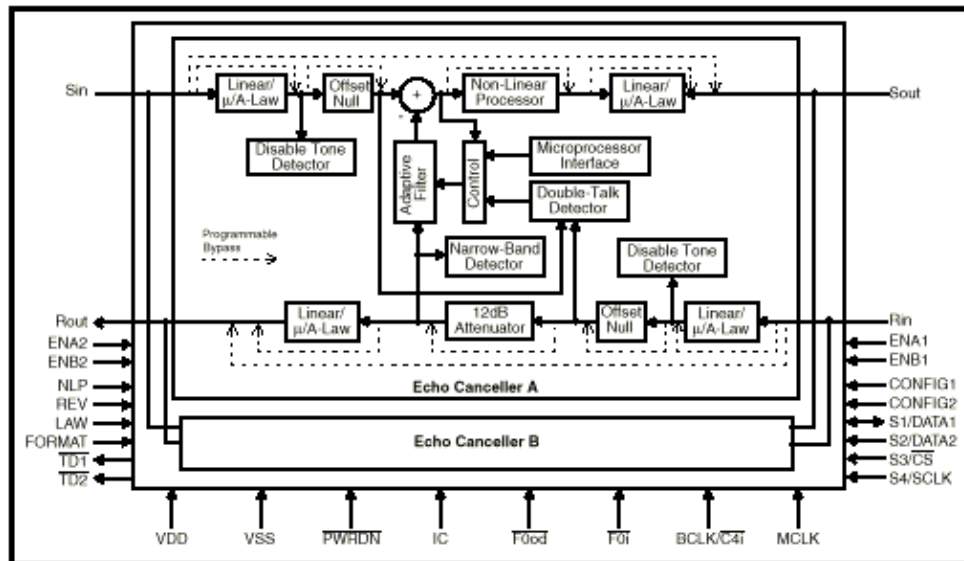
Echo Cancelling

The echo canceller function is provided by the Mitel MT9122 chip located on the UVM back card. It has the following features.

- Adaptive filter for estimating the echo channel
- Subtractor for canceling the echo
- Double talk detector for disabling the filter adaptation during periods of double-talk
- Non Linear Processor for suppression of residual echo
- Disable tone detector for detecting valid disable tones at the input of receive and send paths
- Narrow Band detector for preventing adaptive filter divergence caused by narrow band signals. For example, if dual tone multifrequency (DTMF) tone is present, this may cause the adaptive filter to diverge.
- Offset Null filters for removing the DC component in PCM channels
- 12dB attenuator for signal attenuation
- PCM encoder/decoder compatible with Mu/A law

UVM echo cancellers are always active, but they need to reconverge every time the ERL (or some other characteristic of the echo) changes. It is possible that the ERL changes significantly when the called party goes from ringing state to off-hook state, or goes from talking state to hold state. You can determine the ERL for each of these states by sending DTMF tones and watching Rx/TX levels on the **dchst** screen. A low ERL during the ringing state or hold state would explain the echo heard during those states.

A diagram of the Mitel MT9122 is shown here.



Voice Interworking Service Module

The Voice Interworking Service Module (VISM) uses high performance digital signal processors and dual control processors with advanced software to provide a fully nonblocking architecture that supports the following functions:

- Eight T1 and E1 interfaces per service module, up to 24 service modules per MGX 8800
- Programmable echo cancellation up to 128 msec
- VoIP support
- ATM adaptation layer 1 (AAL1), AAL2, and AAL5 standards support
- Primary Rate Interface (PRI) support
- Fax and modem tone detection for compression and echo cancellation control
- Statistics collection
- Standards-based alarm and fault management
- Simple Network Management Protocol (SNMP) configuration and access
- Redundancy with standby switchover
- Voice Compression
 - ◆ PCM (G.711) support for a total of 192 DS0's per VISM
 - ◆ ADPCM (G.726) support for a total of 145 DS0's per VISM
 - ◆ CS-ACELP (G.729A/B) support for a total of 145 DS0's per VISM
- No R1 and R2 signaling support at this time
- CAS protocol support by mapping the CAS signals to Simple Gateway Control Protocol (SGCP) events

For Release 1.5.04, the VISM supports 2 modes of operation:

- VOIP Switching. In this mode, the VISM functions as a Media Gateway to perform call control in

conjunction with a Call Agent such as the Cisco VSC3000 to provide voice service over existing packet networks.

- AAL2 Trunking. In this mode, the VISM function is similar to the CESM and also offers echo cancellation and G.711/G.726/G.729A/G.729B compression. No Call Agent is required.

AAL2 Trunking and VOIP switching modes are incompatible and can not be implemented on the same VISM. The default mode is VOIP Switching. If AAL2 Trunking mode is selected, the VISM will reset and any existing configuration will be erased. A reset may take as long as five minutes. To view the existing mode, issue the **dspvismparam** command.

```
mgx1.1.11.VISM8.s > dspvismparam
```

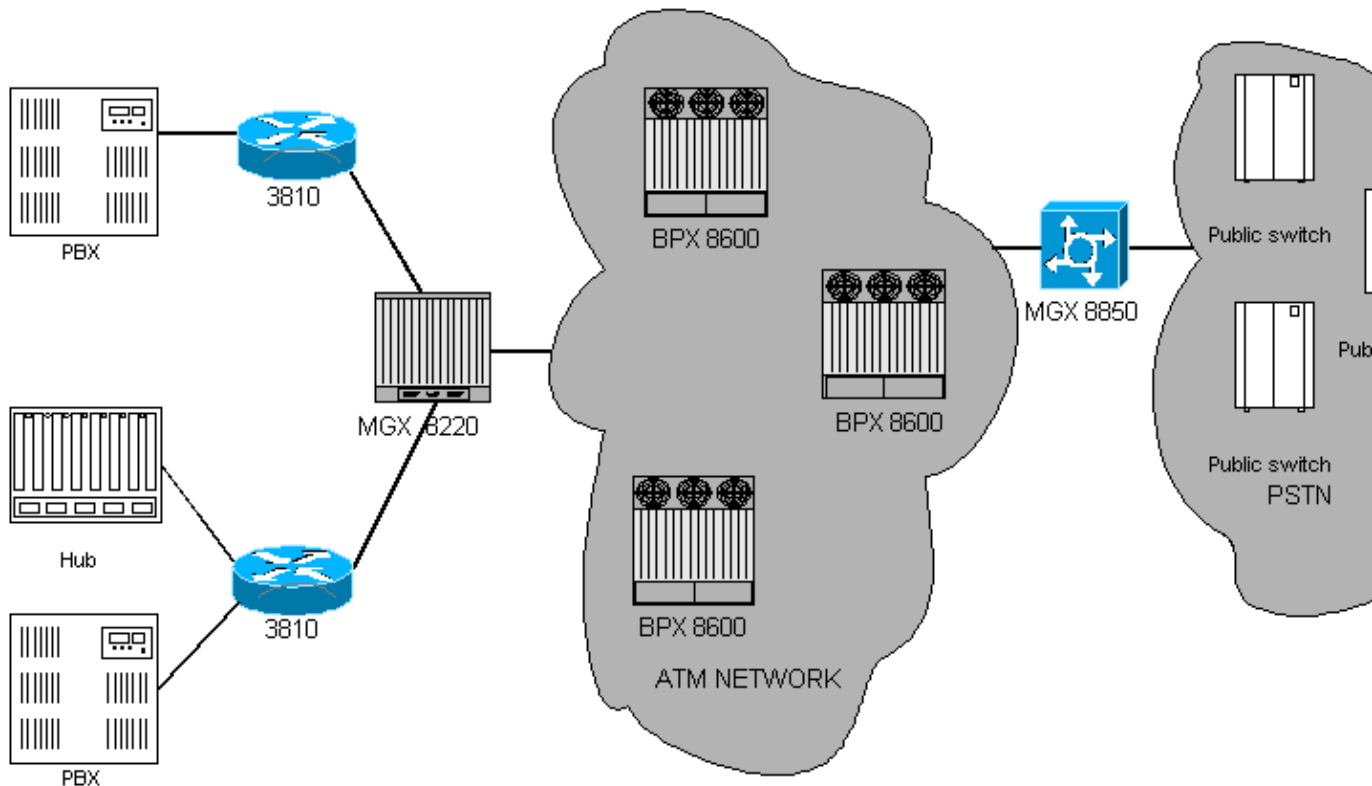
```
VISM mode:                voipSwitching
CAC flag:                  enable
DS0s available:           240
Template number:          2
Percent of functional DSPs: 100
IP address:                0.0.0.0
Subnet mask                0.0.0.0
RTCP report interval:     1000
RTP receive timer:        disable
ControlPrecedence/Tos:    0x60
BearerPrecedence/Tos:     0xa0
Aal2 muxing status:       disable
Tftp Server Dn            TFTPDOMIAN
```

VISM firmware is bundled with MGX 8850 firmware. The VISM release train is different from the MGX 8850 release train. This is reflected in the use of unique identifiers in the firmware image filenames. Once a CCO user has selected the MGX 8850 firmware bundle with the desired VISM release, the images need to be downloaded and unbundled. All registered CCO users who are logged in have access to the Downloads – WAN Switching Software (registered customers only) for firmware downloads.

Clocking is only supported on VISM line 1. The VISM uses line 1 to derive clock used to transmit data for remaining seven T1 lines.

AAL2 Trunking

The VISM supports AAL2 Trunking through the VISM T1 back card or through the SRM T3 interface. Use VoAAL2 as a solution for point-to-point applications as well as for Integrated Voice/Data Access using other Cisco multiservice access products such as the Cisco 2600 series, 3600 series, and MC3810. An application for AAL2 Trunking is illustrated below.



VOIP Switching

The VISM supports VOIP Switching through the VISM T1 back card. VISM operates in conjunction with a Call Agent, such as the Cisco VSC 3000, and two networks are used. The Call Agent connects to the telephone SS7 network and handles the call-control signaling. The VISM connects to an IP network (over an AAL5 PVC) and handles the voice payload between the calling and called parties.

VISM and Call Agent communicate with each other and their activities are coordinated through either SGCP or MGCP. In VOIP Switching mode, the VISM also supports CAS backhaul through data networks. For CAS backhaul, the VISM translates standard trunk signaling protocols to SGCP (MGCP) messages and transmits the messages to the Call Agent. The trunk signaling translation information is stored on the VISM in a file called a CAS variant. Supported CAS variants are:

- E&M Wink Start DID DOD (MF and DTMF)
- E&M Immediate Start DID DOD (DTMF)
- E&M Ground Start (DTMF)

FastPAD Multimedia

The FastPAD Multimedia (FastPADmm) can no longer be ordered. Cisco will support the FastPAD in customer networks until 2003. The following steps describe how to adjust voice level settings at local and remote sites when using the FastPAD Multimedia (FastPADmm).

1. Establish a connection with a person at the remote site.
2. From the Configure menu, select the expansion channel number where the connection is established.
3. Using the <down arrow>, step through the configuration to the point where the In and Out levels are set.
4. Speak to the person at the remote end and get feedback on how you sound to them. If your voice sounds quiet, change the In at your end to be more negative. If your voice sounds loud, change the In at your end to be less negative. Continue adjustments until an appropriate level is reached.

5. Have the person on the other end speak, and adjust the Out parameters at your end accordingly.
6. Repeat this process for each voice channel.
7. Save the settings to active and save the configuration.

MC3810

The Cisco MC3810 is a compact, low-cost multiservice access concentrator that integrates data, voice/fax, and video signals and connects them to Asynchronous Transfer Mode (ATM), Frame Relay, or leased-line networks.

Initial Release

- Seamlessly integrates data, voice, and video
- Leased line, Frame Relay, and ATM service compatible
- Cisco IOS® based multiprotocol routing, bridging and System Network Architecture (SNA)
- Two serial ports for packet data protocols, SNA
- Six analog or 24/30 digital voice ports
- Quality voice compression at 8 kbps (G.729, G.729A) or 32 kbps ADPCM
- Per-call voice switching
- Fax Relay to 9.6 kbps
- Circuit emulation over ATM for video
- Digital Access and Cross connect System (DACS) compatible drop/insert structured trunk option
- Seamless interoperability and management

Release 2.05 (IOS 11.3(1)MA4 and later)

- **Pulse Dialing**

The Cisco MC3810 supports pulse dialing with the following commands: **dial-type pulse**, **timing pulse**, and **timing pulse-inter digit**. These commands are documented in the Cisco MC3810 Multiservice Concentrator Configuration Guide and the Cisco MC3810 Command Reference Guide.

- **Multi-Chassis Hunt Groups**

Hunt group support has been enhanced to route an incoming call to another configured outgoing trunk if it fails to terminate locally because all ports are busy or if the intended outgoing trunk is down or congested. Using multichassis hunt groups, the Cisco MC3810 can hunt between both local dial peers and network dial peers on the terminating or tandem Cisco MC3810. The system hunts among local dial peers first, and then hunts to the network dial peers. A preference order defined with the preference command applies only within the peer group, so all local peers will be searched first, even if a network peer exists with a higher preference.

- **A/B Bit Conditioning**

Cisco IOS Software Release 11.3(1)MA4 supports three new voice port commands: **condition**, **ignore**, and **define**. These commands allow the Cisco MC3810 to recognize and manipulate different ABCD bit combinations such as on-hook and off-hook signals from the PBX.

- **Digit Forwarding**

In previous releases, the Cisco MC3810 that terminated a voice call would only forward digits that exceeded the destination pattern. In Cisco IOS Software Release 11.3(1)MA4, you can control the number of digits forwarded to the telephony interface. This is critical in configuring a hierarchical dial plan.

- **Voice Default Route**

In previous releases, you could not set a voice default route using all wildcards. In Cisco IOS Software Release 11.3(1)MA4, you can set a default voice route for any fixed length dial string using all wildcards with the **destination-pattern** command.

- **Japan and Australia Call Progress Tones**

Modifications have been made to support Japan and Australia call progress tones.

Release 2.1 (IOS Release 12.0)

- **Common Channel Signaling Support for International Applications Including QSIG**

In addition to the Voice switching capabilities currently available in the MC3810 for FXS, FXO, and E&M, the MC3810 supports dynamic calling using the ITU standard QSIG for Common Channel signaling. This implementation supports full voice switching for both T1 (23B+1D) and E1 (30B+1D).

Transparent signaling for CCS PBXs is also supported providing compressed voice over Frame Relay and ATM support for nearly any CCS-based PBX. Bandwidth is allocated to voice calls dynamically using VAD.

These capabilities are backward compatible with systems using the digital voice module currently shipped.

- **End-to-End Networking (IGX 8400 Internetworking)**

The MC3810 can be used to extend high quality of service (QoS) backbones using the Cisco IGX 8400, terminating voice and data on the MC3810 in smaller branch offices. The IGX 8400 provides a scalable, robust head-end solution to voice and data networking using the MC3810. Refer to *Configuring Cisco MC3810-IGX Interworking* for additional information.

- **Full IOS Feature Set Support**

The MC3810 offers the full range of Cisco IOS routing capabilities including IP, X.25, AppleTalk, DecNET, Vines, and others.

- **Robust Applied Telephony Capabilities**

The packet telephony capabilities of the MC3810 have been enhanced to provide a more robust integrated Data/Voice/Video solution for branch locations.

- **Call Detail Record (CDR)**

The MC3810 supports collection and export of call records to a central information base. Information included is time of call, originating port, terminating port, and duration.

- **Multiflex Trunk module with integrated BRI interface**

This module provides all the same functionality as the existing Multiflex module but now provides an additional interface for BRI data backup. The BRI module provides an S/T interface only, which is ideal for European deployment. An inexpensive NT1 can be used to provide connectivity to ISDN services in the US.

- **Facilities Data Link Capabilities on Multiflex Trunk**

This feature provides carriers the full-featured remote management capabilities that they demand from a manageable CSU.

- **OPX Ring-through**

This feature allows a port on the MC3810 to act like an *Off-Premise Extension* to the PBX. When the PBX attempts to make a connection to the remote voice port on an MC3810, OPX ring-through

allows the PBX to re-route the call if there is no answer.

- **Preference-Based Hunt Group**

The multichassis hunt group is enhanced to allow use of the **preference** command for selecting remote dial peers before local dial peers using the priority values. This greatly extends the capability of the product to support on-net to off-net call rerouting and alternate call center applications.

- **G.726 (ADPCM)**

This newly supported vocoder will be more reliable for digit transport in networks with greater hop counts and support lower speed modems (up to 9.6 kbps).

- **Multilength Dial Patterns**

Dial strings of multiple lengths can now be supported in the same network and on the same MC3810.

- **PBX Connectivity**

A large number of PBXs have been integrated with the MC3810.

Traditional PBX Networking

Most PBX networks today use a *tandem* architecture in which all calls are routed through one or more centrally located nodes before reaching their destinations. This approach has several disadvantages:

- It requires many E1/T1 or Fractional E1/T1 lines to support the trunk groups that are needed for tandem connections. This type of trunking is very bandwidth inefficient because traffic must be backhauled through intermediate switching nodes and numerous small trunk groups carry less traffic than a single large trunk group.
- Multiple tandem hops affect voice quality. PBX tandem networks are not amenable to voice compression because speech signals must go through multiple compression/decompression cycles before reaching their final destinations. The result is impaired voice quality, more voice compression cards, more PBX trunk cards, and many signaling channels.
- Separate trunk groups are needed for data and video communications because compression can be applied to only voice connections. As noted, multiple separate trunk groups are less efficient than a single large one.
- PBX feature operation requires signaling channels on each trunk group to carry feature messages between locations. These numerous signaling channels each require 64 kbps of bandwidth.

Voice Network Switching

Cisco Voice Network Switching (VNS) offers the solution to traditional PBX networks. VNS works in conjunction with Cisco IGX 8400 series wide-area switches to provide switched virtual circuits (SVCs) for voice and data transmission over a Cisco wide-area Asynchronous Transfer Mode (ATM) or Frame Relay network. Customers with tandem private branch exchange (PBX) networks realize substantial savings on facility costs, simplified network topology, and improved bandwidth efficiency with a VNS/IGX backbone. In addition, the VNS system architecture is designed to provide scalability for small to very large IGX networks.

Voice Network Switching, in conjunction with standard QSIG and DPNSS common channel signaling protocols, provides direct call-by-call routing for PBX voice, data, and fax connections, enabling this information to be transported across a Cisco WAN with efficiency and economy. It uses one hop routing, which avoids multiple compression/decompression cycles, and removes several shortcomings of existing PBX networks. VNS revolutionizes PBX communications with advanced switching and signaling technology.

Industry standard signaling protocols for ISDN, Frame Relay, and ATM are supported by VNS. The flexibility of the VNS platform also enables Cisco to respond rapidly to changes in networking standards. Key

standards include:

- International Telephony Union (ITU) Recommendation Q.931 for ISDN breakout
- ECMA 300–102, –143 & –165 standards for QSIG PBX networking protocol
- BTNR standard 188 for DPNSS PBX networking protocol

Voice Compression and Enhanced Voice Quality

With VNS, voice calls are compressed once at the origination point and decompressed once at the destination point. Voice quality is improved by eliminating multiple compression/decompression cycles. Improved voice transmission quality can increase the capacity of the network because more aggressive voice compression can be used. For example, 16 kbps voice compression can be used instead of 24 or 32 kbps compression. Network voice capacity can be doubled without incurring additional bandwidth costs while still maintaining acceptable quality. Another benefit of eliminating the multiple compression/decompression cycles is that fewer voice compression processor resources are required. These benefits are leveraged by the comprehensive voice capabilities already available with Cisco IGX 8400 switches.

Voice Activity Detection

Voice Activity Detection (VAD) sends information only when speech energy is present. When a person is not talking, no data is sent. Because each individual speaks only 40 to 50 percent of the time during a typical conversation, VAD can provide a 50 percent savings in bandwidth for voice connections. When combined with ADPCM compression techniques, VNS provides unparalleled efficiency and economy in transporting voice on ATM networks.

CAS Switching

The UVM supports CAS for VNS by converting CAS signaling and DTMF tones to CCS call-control messages.

The converted CCS messages for all channels on the line travel on a regular t-type or a special td-type PVC connection from the UVM to another UVM card. The VNS device can receive the CCS messages from t-type or td-type PVC connections on the signaling channel of CAS-switching UVM cards in the network.

D-Channel Compression

The UVM supports D-channel compression for VNS release 3.1 This feature compresses the signaling traffic between the application UVM and the network (VNS) UVM. D-channel compression reduces the consumed bandwidth from 64 kbps per VNS signaling channel to 16 kbps or less. It applies to CCS lines or CAS lines where the CAS-switching feature is operating. To enable D-channel compression issue the following command:

- **addcon slot_number.line_number.16 <remote_nodename> slot_number.line_number.16 td**

PBX Networking Features

Another advantage of VNS is that it enables users to migrate from a tandem PBX backbone to a Cisco backbone without any change in network functionality.

VNS supports feature operation with the DPNSS and QSIG common channel signaling protocols, which transmit feature messages as well as call setup and tear-down messages. The Cisco IGX 8400 switches pass these messages transparently to the destination PBX, operating as transit PBX nodes. This transit node functionality provides a standards-based pass-through capability that should support all PBXs that conform

accurately to the DPNSS or QSIG protocols. VNS simplifies PBX network design and management. Complex routing tables, multiple trunk groups, and the complicated network topology normally associated with PBX tandem networks are not required. Instead, the Cisco backbone handles these tandem functions more efficiently by dynamically routing all calls directly to their destinations based on a simple dialing plan.

Supported Specifications

QSIG

- VNS appears as QSIG "Transit Node"
- Most PBX supplementary service messages are passed transparently across an IGX network
- Network dialing plan in VNS database (based on E.164)
- E1 Primary Rate Interface connection to IGX CVM
- Industry standard ISDN/QSIG signaling protocol stack:
 - ◆ Layer 2: Q.921
 - ◆ Layer 3: Q.931 call control
 - ◆ European Computer Manufacturers Association (ECMA) 143 basic call control
 - ◆ ECMA 165 generic functional protocol

DPNSS

- VNS appears as DPNSS "Transit Node"
- PBX supplementary service messages passed transparently across IGX network
- Dialing plan in VNS database (based on E.164)
- E1 Primary Rate Interface connection to IGX CVM
- Industry standard DPNSS signaling protocol stack:
 - ◆ Layer 2: DPNSS LAP
 - ◆ Layer 3: DPNSS call control
 - ◆ DPNSS Supplementary Services: PBX Networking features, sections 7–39

Voice over ATM Trunks

In some networks ATM trunks are used to transport voice traffic. To ensure efficient use of ATM trunk bandwidth, IGX configuration changes may be required. If the FastPacket to ATM cell combining values for voice connections are less than those specified in the table below, trunk bandwidth may be wasted. In the least efficient scenario, only one FastPacket (24 bytes) will be transmitted as the payload of one ATM cell (53 bytes). The Service-level command, **cnfcm**, is used to change the FastPacket to ATM cell combining value by adjusting the trunk wait time for a FastPacket. A larger value indicates that the trunk card will wait a longer time for a second FastPacket before sending out the ATM cell. In most cases, changing **cnfcm** significantly increases available trunk bandwidth and minimally increases voice packet delay.

Compression type	cnfcm parameter	Minimum value
P	NTS	22
C32	VOICE	43
a32	NTS	43
c24	VOICE	57
a24	NTS	57
116v	VOICE	81

116	NTS	81
g729r8v or g729ar8v	VOICE	161
g729r8 or g729ar8	NTS	161

PBX Specifics

This section explains how to configure a PBX to simplify the connection to an IGX network. A common problem with new installations is improperly tuned PBXs.

Not addressed in the following sections, but equally important is the PBX clock configuration. The PBX clock source must be exactly the same as the IGX clock source, or the clock sources used by the PBX and IGX must have the same accuracy and stability. Equivalent clocks are required to keep the slip rate between the PBX and IGX at an acceptable level.

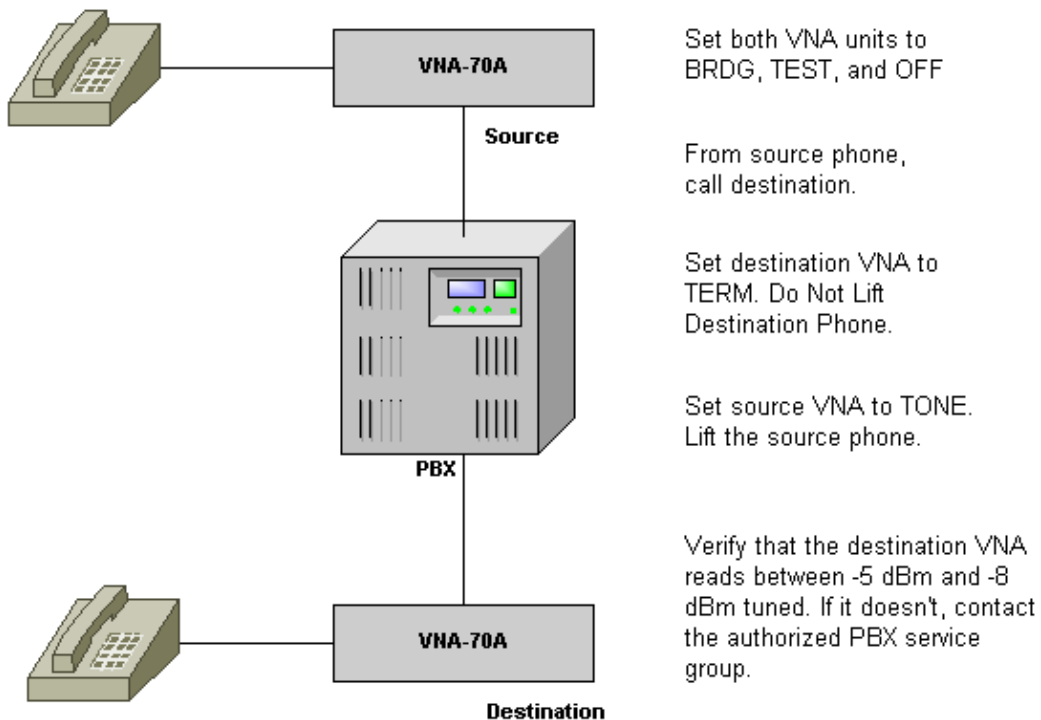
Note: When you add connections between PBXs located in geographically diverse locations, the `cnfclnsigparm <slot>` command may be required to adjust for delay.

Generic PBX Tuning

If voice connections are routed over tandem switches, the PBX network must be tuned to avoid signal loss and degradation. Loss plans are covered in detail in International Telephony Union (ITU) Recommendation G.171 and ANSI TIA/EIA TSB 32 . The loss must be assessed at a number of locations to obtain the worst case scenario when designing a network. The method for tuning PBXs described below requires:

- Two devices capable of generating a 1004 Hz tone at 0 dBm. For example, two MetroTel Voice Network Analyzers (Model VNA-70A).
- Two analog phones.

Follow the procedure in the following diagram.



iSDX

The Integrated Services Digital Exchange (iSDX) is a family of digital PBXs designed for business requirements of 30 to 3000 extensions. There are four iSDX systems: iSDX-T, iSDX-L, iSDX-S, and iSDX Micro. All iSDX systems are based on a common hardware platform and common software. Over 17,500 iSDX systems have been sold in more than 40 countries worldwide, incorporating in excess of 5.5 million lines.

The iSDX is at the forefront in the development of DPNSS and has the highest level of compliance to this standard of any PBX. iSDX is the benchmark with which all other PBXs must interwork.

A caveat with analog voice connections on the iSDX is that the default connection type is SSDC5a, which is a variant of E&M Type V signaling. A caveat with power and ground is that a nonstandard pin from the iSDX allows the passing of ground between the PBX and the Cisco equipment. The iSDX does not use the electrical earth that is supplied with the AC PSU. There is a specific external earthing point for this purpose. There is no information about problems with the DC version.

Meridian

When attaching to a Northern Telecom Meridian with an analog trunk configured for E&M TIE Line, one of the trunk parameters is CPAD. The choices to configure this parameter are:

- C OUT, which sets the trunk's input and output levels to 0 dBm. This is the default setting.
- C IN, which sets the trunk's input level to +7dB and output level to -16 dBm.

The C OUT setting is preferred. The C IN setting results in low volume from the PBX, which affects IGX VAD and onboard echo cancellation.

MD110

The Ericsson MD110 uses a proprietary protocol called System Link that needs 256 kbps of bandwidth for remote PBX downloading. The 256 kbps of bandwidth is distributed over timeslots 1, 3, 5, and 7. These timeslots must be configured as transparent connections on the IGX (for example, no voice compression can be used). There is no transparent connection requirement if QSIG or another CCS-type protocol is used.

PBX Clearing Codes

The table below shows a list of the clearing codes transmitted between PBXs.

Cause	Clearing / Rejection Cause	Cause	Clearing / Rejection Cause
0	Number not obtainable	26	Message not understood
1	Address incomplete	27	Signaling system incompatible (DPNSS) Reserved (DASS2)
2	Network termination	28	Reserved
3	Service Unavailable	29	Transferred (DPNSS) Reserved (DASS2)
4		30	NAE error

	Subscribers incompatible		
5	Subscriber changed number	31	No reply from subscriber
6	Invalid request for supplementary service	32	Service termination
7	Congestion	33–34	Reserved
8	Subscriber engaged	35	Out of service
9	Subscriber out of service	36–40	Reserved
10	Incoming calls barred	41	Access barred
11	Outgoing calls barred	42–44	Reserved
12–17	Reserved	45	DTE controlled not ready
18	Remote procedure error	46	DCE controlled not ready
19	Service incompatible	47	Reserved
20	Acknowledge (DPNSS)	48	Subscriber call termination
21	Signal not understood	49	Reserved
22	Invalid signal (DPNSS)	50	ET isolated
23	Service temporarily not available (DPNSS)	51	Local Procedure error
24	Facility not registered (DPNSS) Reserved (DASS2)	52–255	Reserved
25			

References

- *British Telecommunications Network Requirement (BTNR) No. 188. DPNSS Specification.*
- *ECMA 143 QSIG Specification. 2nd edition, December, 1992. An electronic copy can be found at <http://www.ecma-international.org/>.*
- *Cisco IGX 8400 Series Reference*

Related Information

- **Voice Technologies**
- **Cisco AS5300 Universal Access Server**
- **Cisco MC3810 Multiservice Concentrator Configuration Guide**
- **Troubleshoot Voice with TAC Case Collection (registered customers only)**
- **Cisco WAN Switching Solutions – Cisco Documentation**
- **Guide to New Names and Colors for WAN Switching Products**
- **Downloads – WAN Switching Software (registered customers only)**

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