

IP Communications High–Density Digital Voice/Fax Network Module

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Introduction

This document provides a sample configuration for the IP Communications High–Density Digital Voice/Fax Network Module feature, which supports high–density digital voice and low–density analog voice connectivity along with data and integrated access connectivity. The network modules offer built–in T1/E1 ports, and include a single voice interface card (VIC)/voice WAN interface card (VWIC) slot for Foreign Exchange Station (FXS), Foreign Exchange Office (FXO), E&M, software–configured Centralized Automatic Message Accounting (CAMA), direct inward dialing (DID), BRI, or E1 and T1 cards, up to a maximum of four T1/E1 ports. The network modules also support up to 32 HDLC channels with an aggregate capacity of 2.048 Mbps.

Note: The CAMA card (VIC–2CAMA) is not supported. However, any port on the VIC2–2FXO and the VIC2–4FXO can be software configured to support analog CAMA for dedicated E–911 services (North America only).

Symptoms

You can potentially encounter the following symptoms or error messages when configuring IP Communications High–Density Digital Voice/Fax Network Module:

- % No DSP resources available to configure pri–group on controller T1
- %XCCTSP_VOICE–3–NOSDB: No signaling data block is available to build the voice interface (1/0:23) or DSP may not be present

The above errors can be resolved by doing DSP Resource Sharing or adding more digital signal processors (DSPs). For more information refer to the DSP Calculator tool.

Prerequisites

Requirements

Ensure that you meet these requirements before you attempt this configuration:

- Be aware that software echo cancellation is the default configuration -- G.168-compliant echo cancellation is enabled by default with a coverage of 64 milliseconds.
- Be aware that only Packet Fax/Voice DSP modules (PVDM2s) are supported.
- Use only voice interface cards that start with VIC2, except for VIC-1J1, VIC-2DID, and VIC-4FXS/DID.
- Be aware that the DID feature in VIC-4FXS/DID is not supported in the original release of this feature. However, the DID feature in VIC-4FXS/DID is supported starting with Cisco IOS Release 12.3(14)T.
- The CAMA card (VIC-2CAMA) is not supported. Any port on the VIC2-2FXO and the VIC2-4FXO can be software configured to support analog CAMA for dedicated E-911 services (North America only).

Components Used

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

- IP Plus image (minimum) of Cisco IOS Release 12.3(7)T or a later release. Cisco IOS Release 12.3(14)T is required for the DID feature on the VIC-4FXS/DID card.
- In a Cisco CallManager network, the CCM 4.0(1) SR1 or CCM 3.3(4) release must be installed.
- Cisco 2600XM, Cisco 2691, Cisco 3600 series, Cisco 2800, and Cisco 3800

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

Refer to the Cisco Technical Tips Conventions for more information on document conventions.

Background Information

The IP Communications High-Density Digital Voice/Fax Network Module feature supports high-density digital voice and low-density analog voice connectivity along with data and integrated access connectivity. This section provides information about the following:

- Key Features
- Hardware Configuration Options
- DSP Resource Sharing
 - ◆ Search Order for DSPs
 - ◆ Codec Combinations for DSP Sharing
- Single-Point Configuration of MGCP Voice Gateways in AVVID Networks
- Network Clock Timing

- Configuring Drop and Insert

Key Features

The IP Communications High-Density Digital Voice/Fax Network Module features the following:

- High-density digital voice connectivity up to 4 T1/E1 ports or 120 medium-complexity channels
- High-density data WAN connectivity up to 4 T1/E1 ports
- Analog voice connectivity up to 4 ports
- Built-in T1/E1 ports configurable for either T1 or E1 operation via command-line interface (CLI)
- Up to 32 HDLC channel groups with a total bandwidth of 2.048 Mbps
- PVDM2 technology that supports higher call densities and more flexibility in channel allocation per DSP
- G.168-compliant echo cancellation for tail circuits up to 64 milliseconds

Hardware Configuration Options

The IP Communications High-Density Digital Voice/Fax Network Module is available on three network modules, with the option of zero, one, or two built-in T1/E1 ports.

Each built-in port can be software-configured to support either T1 or E1 operation. However, if you configure two onboard controllers, both must be T1 or both must be E1. Each network module also supports a single VIC/VWIC slot that can be fitted with a Cisco VWIC or Cisco VIC. The Cisco VICs are daughter cards that install into the network modules and provide the interface to the PSTN and to telephony equipment (PBX, key systems, fax machines, and phones). The Cisco VWICs are daughter cards that provide the interface to the PBX, PSTN, and WAN.

The IP Communications High-Density Digital Voice/Fax Network Module must be used with the new PVDM2s, providing scalability of 4 to 120 channels using the latest digital signal processing technology. Up to a maximum of four PVDM2s can be installed on each NM-HDV2 network module. You can select the minimum number and density-type PVDM2s depending on the voice channels currently needed, and then scale the number of PVDMs as requirements expand. These new PVDM2 SIMMs can be configured for high complexity, medium complexity, or flex. Flex complexity is the default configuration. In this mode, the network module will dynamically select the appropriate codec (medium or high) depending on the PVDM2s available. In addition, the DSPs on the PVDM2s can be shared across multiple IP Communications High-Density Digital Voice/Fax Network Modules installed on a voice gateway router. The following list summarizes the configuration options. The table following the list summarizes the numbers of channels (based on complexity) for the PVDM2. On the Cisco 2800 series and Cisco 3800 series Integrated Services Routers, the IP Communications High-Density Digital Voice/Fax Network Module can be used with PVDM2s on the motherboard of the platform.

Network Modules:

- NM-HDV2 --- 1-slot IP communications voice/fax network module
- NM-HDV2-1T1/E1 --- 2-slot IP communications voice/fax network module with one slot for T1/E1 interface
- NM-HDV2-2T1/E1 --- 2-slot IP communications voice/fax network module with two slots for T1/E1 interface

Packet Voice Data Modules:

- PVDM2-8 --- 8-channel packet fax/voice DSP module
- PVDM2-16 --- 16-channel packet fax/voice DSP module

- PVDM2-32 --- 32-channel packet fax/voice DSP module
- PVDM2-48 --- 48-channel packet fax/voice DSP module
- PVDM2-64 --- 64-channel packet fax/voice DSP module

VIC and VWIC Options:

- VIC2-2FXO --- 2-port voice interface card FXO (universal) also supports CAMA
- VIC2-4FXO --- 4-port VIC FXO (universal) also supports CAMA
- VIC2-2FXS --- 2-port VIC FXS
- VIC-4FXS/DID --- 4-port FXS or DID VIC
- VIC2-2E/M --- 2-port voice interface card E&M
- VIC2-2BRI-NT/TE --- 2-port voice interface card BRI
- VIC-2DID --- 2-port DID voice/fax interface card
- VIC-1J1 --- 1-port J1 voice interface card
- VWIC-1MFT-T1 --- 1-port RJ-48 multiflex trunk T1
- VWIC-2MFT-T1 --- 2-port RJ-48 multiflex trunk T1
- VWIC-2MFT-T1-D1 --- 2-port RJ-48 multiflex trunk T1 with drop and insert
- VWIC-1MFT-E1 --- 1-port RJ-48 multiflex trunk E1
- VWIC-2MFT-E1 --- 2-port RJ-48 multiflex trunk E1
- VWIC-2MFT-E1-D1 --- 2-port RJ-48 multiflex trunk E1 with drop and insert
- VWIC-1MFT-G703 --- 1-port RJ-48 multiflex trunk G.703
- VWIC-2MFT-G703 --- 2-port RJ-48 multiflex trunk G.703

Table 1 Channel Availability for PVDM2 Modules Based on Codec Complexity

Network Module	Max	High	Medium	Flex
	DSPs	Complexity	Complexity	Complexity
PVDM2-8	1	4	4	8
PVDM2-16	1	6	8	16
PVDM2-32	2	12	16	32
PVDM2-48	3	18	24	48
PVDM2-64	4	24	32	64

DSP Resource Sharing

When one IP Communications High-Density Digital Voice/Fax Network Module does not have enough DSP resources, it can use DSPs from other NM-HDV2s on the same router, or DSPs available on the motherboard of the Cisco 2800 series and Cisco 3800 series Integrated Services Routers. This is referred to as DSP sharing. By default, the NM-HDV2s and the onboard PVDM2 DSPs on the Cisco 2800s and Cisco 3800s are configured for "no sharing," and must be turned on to share or export their resources. An NM-HDV2 that needs to import DSPs does not need any special configuration.

Search Order for DSPs

All available DSPs that are configured for sharing are pooled together in the search. An NM-HDV2 without any DSP resources will begin searching first on the motherboard (supported only on the Cisco 2800 and Cisco 3800 platforms), followed by other NM-HDV2 modules. Network modules are searched according to slot number. The **network-clock participate** command must be configured on the network modules that are sharing resources and need DSP resources.

Codec Combinations for DSP Sharing

When network modules or PVDM2s on the motherboard are configured for DSP sharing, the codec complexity has to match. A local resource sharing or importing from a remote network module must match its characteristics, that is, a high-complexity network module can only share from another high-complexity network module, whereas a flex-complexity network module can share DSPs from both high-complexity and flex-complexity network modules. The following table summarizes the codec combinations for DSP-sharing.

Table 2 Codec Complexity Settings for DSP Resource Sharing Between Local and Remote Sources

Local DSP resource (import)	Remote DSP resource (export)		
	High complexity	Medium complexity	Flexible complexity
High complexity	yes	no	no
Medium complexity	yes	yes	no
Flexible complexity	yes	no	yes

Single-Point Configuration of MGCP Voice Gateways in AVVID Networks

When using a Cisco IOS voice gateway in conjunction with MGCP and Cisco CallManager, you can complete the necessary configuration for a given gateway on the Cisco CallManager server and download the configuration to that gateway through a TFTP server. To enable this configuration on the NM-HDV2 modules, the **card type** command must be used first:

```
card type {t1 | e1} slot subslot
```

Network Clock Timing

Voice systems that pass digitized (pulse code modulation or PCM) speech have always relied on the clocking signal being embedded in the received bit stream. This reliance allows connected devices to recover the clock signal from the bit stream, and then use this recovered clock signal to ensure that data on different channels keep the same timing relationship with other channels.

If a common clock source is not used between devices, the binary values in the bit streams may be misinterpreted because the device samples the signal at the wrong moment. As an example, if the local timing of a receiving device is using a slightly shorter time period than the timing of the sending device, a string of 8 continuous binary 1s may be interpreted as 9 continuous 1s. If this data is then re-sent to further downstream devices that used varying timing references, the error could be compounded. By ensuring that each device in the network uses the same clocking signal, you can ensure the integrity of the traffic.

If timing between devices is not maintained, a condition known as clock slip can occur. Clock slip is the repetition or deletion of a block of bits in a synchronous bit stream due to a discrepancy in the read and write rates at a buffer.

Slips are caused by the inability of an equipment buffer store (or other mechanisms) to accommodate differences between the phases or frequencies of the incoming and outgoing signals in cases where the timing of the outgoing signal is not derived from that of the incoming signal.

A T1 or E1 interface sends traffic inside repeating bit patterns called frames. Each frame is a fixed number of bits, allowing the device to see the start and end of a frame. The receiving device also knows exactly when to expect the end of a frame simply by counting the appropriate number of bits that have come in. Therefore, if the timing between the sending and receiving device is not the same, the receiving device may sample the bit stream at the wrong moment, resulting in an incorrect value being returned.

Even though Cisco IOS software can be used to control the clocking on these platforms, the default clocking mode is effectively free running, meaning that the received clock signal from an interface is not connected to the backplane of the router and used for internal synchronization between the rest of the router and its interfaces. The router will use its internal clock source to pass traffic across the backplane and other interfaces.

For data applications, this clocking generally does not present a problem as a packet is buffered in internal memory and is then copied to the transmit buffer of the destination interface. The reading and writing of packets to memory effectively removes the need for any clock synchronization between ports.

Digital voice ports have a different issue. It would appear that unless otherwise configured, Cisco IOS software uses the backplane (or internal) clocking to control the reading and writing of data to the DSPs. If a PCM stream comes in on a digital voice port, it will obviously be using the external clocking for the received bit stream. However, this bit stream will not necessarily be using the same reference as the router backplane, meaning the DSPs will possibly misinterpret the data that is coming in from the controller.

This clocking mismatch is seen on the router's E1 or T1 controller as a clock slip—the router is using its internal clock source to send the traffic out the interface but the traffic coming in to the interface is using a completely different clock reference. Eventually, the difference in the timing relationship between the transmit and receive signal becomes so great that the controller registers a slip in the received frame.

To eliminate the problem, change the default clocking behavior through Cisco IOS configuration commands. It is absolutely critical to set up the clocking commands properly.

Even though these commands are optional, we strongly recommend you enter them as part of your configuration to ensure proper network clock synchronization:

- **network-clock-participate** [*slot slot number* | **wic** *wic-slot* | **aim** *aim-slot-number*]
network-clock-select *priority* { **bri** | **t1** | **e1** } *slot/port*

The **network-clock-participate** command allows the router to use the clock from the line via the specified slot/wic/aim and synchronize the onboard clock to the same reference.

If multiple VWICS are installed, the commands must be repeated for each installed card. The system clocking can be confirmed using the **show network clocks** command.



Caution: If you are configuring a Cisco 2600 XM voice gateway with an NM-HDV2 or

NM-HD-2VE installed in slot 1, do not use the **network-clock-participate slot 1** command in the configuration. In this particular hardware scenario, the **network-clock-participate slot 1** command is not necessary. If the **network-clock-participate slot 1** command is configured, voice and data connectivity on interfaces terminating on the NM-HDV2 or NM-HD-2VE network module may fail to operate properly. Data connectivity to peer devices may not be possible at all, and even loopback plug tests to the serial interface spawned via a channel-group configured on the local T1/E1 controller will fail. Voice groups such as CAS ds0-groups and ISDN pri-groups may fail to signal properly. The T1/E1 controller may accumulate large amounts of timing slips as well as Path Code Violations (PCVs) and Line Code Violations (LCVs).

Configuring Drop and Insert

T1/E1 VWICs with drop-and-insert functionality connect other devices to a T1 or E1 data stream. Drop-and-insert technology is sometimes called TDM cross-connect.

This feature supports both inter-network-module and intra-network-module drop and insert. If you configure inter-network-module drop and insert, you must also configure network clocking.

Note: If you will be configuring drop and insert, the T1 or E1 framing under the controllers involved (where the tdm-groups are configured), needs to be the same. If different framing types are used, the signaling bits may not be understood properly when a channel from one controller is dropped and inserted into a channel from another controller.

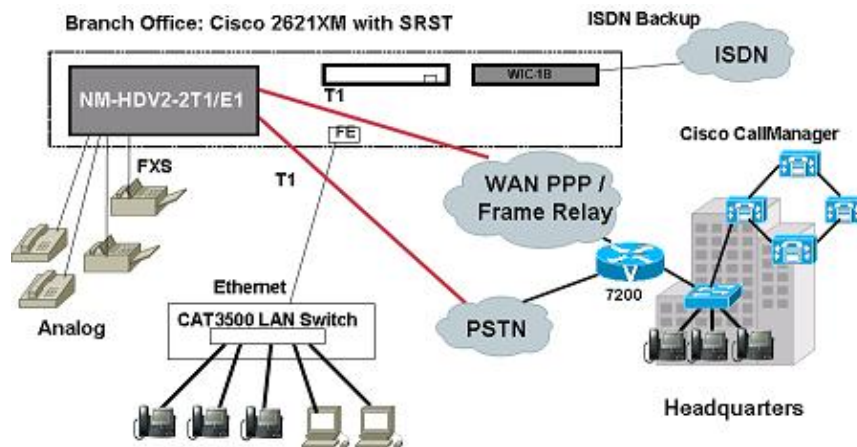
Configure

In this section, you are presented with the information to configure the features described in this document.

Note: Use the Command Lookup Tool (registered customers only) to obtain more information on the commands used in this section.

Network Diagram

This document uses this network setup:



Configurations

This document uses these configurations:

- Channel Bank Support
- Normal VoIP Showing Some Calls
- MGCP Configuration
- Fax Relay Configuration

Channel Bank Support

```
!  
card type t1 3 1  
!  
!  
controller T1 3/0
```

```

framing esf
linecode b8zs
ds0-group 0 timeslots 1 type fxo-loop-start
ds0-group 1 timeslots 2 type fxo-loop-start
!
!
connect test_1 voice-port 3/0/0 T1 3/0 0
!
!
connect test_2 voice-port 3/0/1 T1 3/0 1
!
!
!
voice-port 3/0/0
  signal loopstart
  description FXS LoopStart Port
!
voice-port 3/0/1
  signal loopstart
  description FXS LoopStart Port
!

```

Normal VoIP Showing Some Calls

Originating Side

```

!
card type t1 2 1
!
controller T1 2/0
framing esf
linecode b8zs
ds0-group 0 timeslots 1-24 type e&m-immediate-start
!
dial-peer voice 4100 pots
destination-pattern 4100
port 2/0:0
!
dial-peer voice 999 voip
destination-pattern 99..
session target ipv4:11.3.14.25
codec gsmfr
!

```

Terminating Side

```

!
card type t1 1 1
!
controller T1 1/0
framing esf
clock source internal
linecode b8zs
ds0-group 0 timeslots 1-24 type e&m-immediate-start
!
dial-peer voice 999 pots
destination-pattern 99..
port 1/0:0

```



```
!  
dial-peer voice 1111 voip  
incoming called-number 99..  
codec gsmfr  
!
```

MGCP Configuration

```
!  
card type t1 4 1  
!  
controller T1 4/0  
framing esf  
linecode b8zs  
ds0-group 0 timeslots 1-24 type e&m-immediate-start  
!  
mgcp  
mgcp call-agent 10.1.0.60 service-type mgcp version 0.1  
mgcp package-capability rtp-package  
no mgcp package-capability atm-package  
no mgcp package-capability res-package  
mgcp fax t38 inhibit  
!  
ccm-manager mgcp  
!  
!--- Required for fall back  
  
call app alternate default  
!  
dial-peer voice 4000 pots  
application mgcpapp  
port 4/0:0!
```

Fax Relay Configuration

Global Configuration for Fax Pass-Through

```
voice service voip  
fax protocol passthrough g711ulaw
```

Dial-Peer Level Configuration for Fax Pass-Through

```
dial-peer voice 300 voip  
destination-pattern 93...  
session target ipv4:1.3.28.103  
fax rate disable  
fax protocol passthrough g711ulaw
```

Global Configuration for Fax Relay

```

voice service voip

!--- this line will not show as it is default setting

fax protocol cisco

Dial-Peer Level Configuration for Fax Relay

dial-peer voice 300 voip
destination-pattern 93...
session target ipv4:1.3.28.103

!--- this line will not show as it is default setting

fax protocol cisco

Global Configuration for T.38

voice service voip
fax protocol t.38

Dial-Peer Level Configuration for T.38

dial-peer voice 300 voip
destination-pattern 93...
session target ipv4:1.3.28.103
fax protocol t38

```

Verify

Use this section to confirm that your configuration works properly.

The Output Interpreter Tool (registered customers only) (OIT) supports certain **show** commands. Use the OIT to view an analysis of **show** command output.

Issue the following **show connection** command to verify that the E&M port 2/0 is configured for a channel bank connection with time slot 1 on T1 1/0.

```

Router#show connection ?

all                All Connections
elements          Show Connection Elements
id                ID Number
name              Connection Name
port              Port Number

Router#show connection all

ID                Name                Segment 1                Segment 2                State
=====

```

Troubleshoot

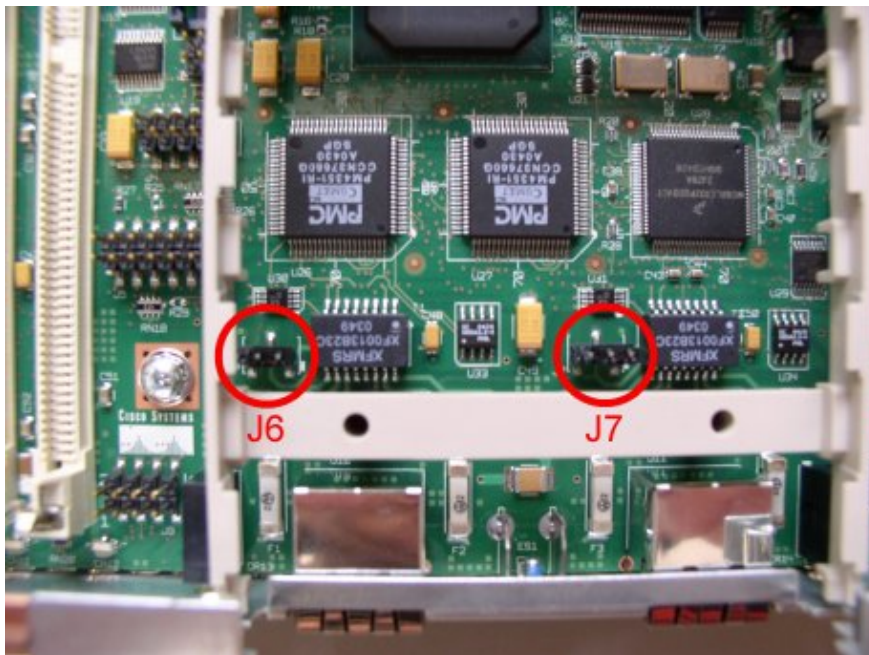
This section provides information you can use to troubleshoot your configuration.

Troubleshooting Procedure

This is troubleshooting information relevant to the E1 card type.

When the onboard controllers are configured for E1 mode, it is possible that the E1 controllers may not come up properly even when connected to known good E1 Telco lines. The output of the **show controllers E1** command can indicate large accumulations of Line Code Violations (LCVs) and Path Code Violations (PCVs). The problem can be the result of how the E1 line has been provisioned by the Telco; specifically whether Wet Current is provided or not.

1. On the NM–HDV2 product, there are two jumper blocks that control whether the onboard T1/E1 controllers support Wet Current or not. These jumpers are identified on the Printed Circuit Board (PCB) of the Network Module as J6 and J7 (see photograph). J6 is the jumper block for onboard controller 1 while J7 is the jumper block for onboard controller 0. The pin count for each jumper block is from 1 to 3. Pin 1 is the rightmost pin and Pin 3 is the leftmost pin. Current production NM–HDV2s now ship with the jumper blocks set for Normal Mode.



2. When Pins 1 and 2 are short-circuited (Right Jumper Setting) the onboard controller is set for "Wet Current Mode", and when Pins 2 and 3 are short-circuited (Left Jumper Setting) the onboard controller is set for "Normal Mode". Early production NM–HDV2s shipped with the jumper blocks set to expect Wet Current to be supplied by the Telco, and this causes problems for some E1 lines.
3. When you move the setting to Normal Mode, it typically clears up the problem.

Related Information

- [Network Module Installation](#)
 - [Technical Support & Documentation – Cisco Systems](#)
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