

DSP Allocation on a Cisco ICS 7750

Document ID: 7123

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Introduction

This document describes issues related to Digital Signal Processors (DSPs), codecs, and transcoding on Analog Station Interface (ASI) and Multiservice Route Processor (MRP) Voice Interface Cards (VICs) installed in the Cisco ICS 7750.

Prerequisites

Requirements

This document applies to Cisco ICS 7750 with software version 2.1.x or later. This document does not detail any of the tasks required to properly administer or use the following:

- Cisco CallManager 3.1
- Cisco Unity 3.0
- Cisco Voice Gateways
- Cisco Voice over IP Solutions

You should have a basic understanding of the above components. A basic understanding of Cisco ICS 7750 hardware is also helpful. For more information regarding ICS 7750 hardware, please refer to the Cisco ICS 7750 Hardware Installation Guide and the Cisco ICS 7750 Administration and Troubleshooting Guide.

Components Used

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

- Software revision Bundle 2.1.0 or later

- Hardware revision MRP 200, ASI160 and ASI81

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, see the Cisco Technical Tips Conventions.

DSP Allocation

There are two DSP modules (PVDM) for each MRP or ASI card (MRP has two SIMM Slots). Each PVDM can support 1 to 5 DSPs; therefore, it only can have a maximum of 10 DSPs for each MRP or ASI card.

The following table shows how many DSPs certain PVDMs can support.

Part Number	# of DSPs
PVDM-4	1 DSP
PVDM-8	2 DSPs
PVDM-12	3 DSPs
PVDM-16	4 DSPs
PVDM-20	5 DSPs

Determining How Many DSPs are Needed

The number of DSPs needed for each voice interface depends on the following two factors:

- Codec Complexity
- Type of Codec Selected

The table below shows how to determine the number of DSPs needed for each channel.

	Transcoding (XCODE)	High Complexity Analog (FIXHC)	Medium Complexity Analog (FIXMC)	High Complexity Digital (FLEX6)	Medium Complexity Digital (FLEX8)
Codec (Channels/DSP)	G.711 (2)	G.711	G.711	G.711	G.711
	G.726 (2)	(2) G.726	(4) G.726	(6) G.726	(8) G.726
	G.729 (2)	(2) G.729	(4) G.729	(3) G.729a	(4) G.729a(b)
	G.723.1	(2) G.723.1	(4)	(3) G.723.1	(4)

	(2)	(2)		(2)	
	n/a	Fax Relay	Fax Relay	Fax Relay	n/a
		(2)	(4)	(3)	
Cards Supported	All 2-port Analog VICs	8 Ports and 16 Ports on ASI	8 Ports and 16 Ports on ASI	All Digital VWICs	n/a

Analog VICs VIC-2DID, VIC-2DID, VIC-2FXS, VIC-2FXO, VIC-2-FXO-M1, VIC-2-FXO-M2, VIC-2FXO-M3 , VIC-2E/M

Basic Rate Interface (BRI) VICs Use same image as Analog VICs

Digital VWICs VWIC-1MFT-T1, VWIC-1MFT-E1, VWIC-2MFT-T1, VWIC-2MFT-E1.

Choosing a DSP Firmware Image

When choosing DSP firmware, it is important to consider the following factors:

- The codecs that need to be supported
- The number of voice channels required for each DSP
- Echo cancellation coverage

The table below summarizes the various DSP firmware images. The acronyms FIXHC, FIXMC, FLEX6, FLEX8, and XCODE are used in the output of the **show voice dsp** command.

```
MRP-00019666360f# show voice dsp
```

```

          BOOT          PAK
TYPE DSP CH  CODEC   VERS STATE  STATE  RST AI PORT TS ABORT  TX/RX-PAK-CNT
=====
Current total analog signaling channels: 2
Current max allowed digital timeslot for voice: 0
Current number of DSP group: 1

Group 0:
  Current allocated analog signaling channels: 2
  Current free analog signaling channels: 0
  Current allocated digital signaling channels: 0
  Current free digital signaling channels: 0
  Port type: ANALOG
  Port(s) served: 0/0 0/1
SPMM DSPRM State Image D-sig D-sig A-sig A-SIG Mips Voice/Xcode
Dsp Dsp          UP   FIXED 0    0    2    0    Free Chan
0/0 0           UP   FIXED 0    0    2    0    N/A 0

```

Configuring Codec Complexity

Card Type	Image Supported	Default Image
8-port and 16-port FXS in an ASI 81 or ASI 160	Medium and High	Medium
2-port Analog VICs	Complexity High Complexity	Complexity High Complexity
BRI VICs	High Complexity	

		High Complexity
MFT-T1 and MFT-E1 cards	Medium and High Complexity	High Complexity

Syntax **codec complexity** [**High** | **Medium**]

- When there are two MFT-T1 or MFT-E1 cards installed in the MRP, choosing DSP firmware for one MFT-T1 or MFT-E1 forces the same firmware to be used on the other MFT-T1 or MFT-E1 in the MRP.
- To ensure that there are no active calls on the voice card before changing codec complexity, all ds0-group and pri-group configurations must be removed from T1 or E1 cards and all voice ports must be shut down on analog voice cards. For an ASI 160, all 16 ports can be shut down at the same time by entering the **shutdown** command in voice card configuration mode. For example, by entering the command **voice-card 0**, then the command **shutdown**, all voice ports on the analog voice card in slot 0 are shut down.

Determining DSP Resources Available on an MRP or ASI

Using the show diag Command

The **show diag** command shows the number of DSPs that are available.

```
ASI81# show diag
```

```
Slot 0:
```

```
ICS7750-MRP200 Mainboard Port adapter, 10 ports
Port adapter is analyzed
Port adapter insertion time unknown
EEPROM contents at hardware discovery:
Hardware Revision      :6.0
PCB Serial Number     :JAD0442076D
Part Number           :73-4341-06
Board Revision        :B0
Fab Version           :04
EEPROM format version 4
EEPROM contents(hex):
0x00: 04 FF 40 01 A8 41 06 00 C1 8B 4A 41 44 30 34 34
0x10: 32 30 37 36 44 82 49 10 F5 06 42 42 30 02 04 FF
0x20: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x30: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x40: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
```

```
!--- PVDM slot 0, corresponding to VIC slot 0.
```

```
Packet Voice DSP Module Slot 0:
Hardware Revision      :2.2
Part Number           :73-4870-02
Board Revision        :01
Deviation Number      :0-0
Fab Version           :02
PCB Serial Number     :ICP04260030
RMA Test History      :00
RMA Number            :0-0-0-0
RMA History           :00
Processor type        :D2
```

!--- Slot contains a PVDM-16.

Number of DSPs :4
DSP memory size(in kwords):256
Type of DSP :Unknown (210)
EEPROM format version 4
EEPROM contents(hex):
0x00: 04 FF 40 01 5D 41 02 02 82 49 13 06 02 42 30 31
0x10: 80 00 00 00 00 02 02 C1 8B 49 43 50 30 34 32 36
0x20: 30 30 33 30 03 00 81 00 00 00 00 04 00 09 D2 FF

!--- PVDM slot 1, corresponding to VIC slot 1.

Packet Voice DSP Module Slot 1:
Hardware Revision :2.2
Part Number :73-4793-03
Board Revision :A0
Deviation Number :0-0
Fab Version :03
PCB Serial Number :ICP045300RP
RMA Test History :00
RMA Number :0-0-0-0
RMA History :00
Processor type :02

!--- Slot contains a PVDM-20.

Number of DSPs :5
DSP memory size(in kwords):256
Type of DSP :TMS320C549
EEPROM format version 4
EEPROM contents (hex):
0x00: 04 FF 40 01 D7 41 02 02 82 49 12 B9 03 42 41 30
0x10: 80 00 00 00 00 02 03 C1 8B 49 43 50 30 34 35 33
0x20: 30 30 52 50 03 00 81 00 00 00 00 04 00 09 02 FF

WIC Slot 0:

8 Port FXS Voice Interface Module WAN daughter
card-Version 4 TLV Cookie Format
Hardware Revision :1.0
Part Number :73-6755-01
Board Revision :01
Deviation Number :0-0
Fab Version :01
PCB Serial Number :BENNY012345
RMA Test History :00
RMA Number :0-0-0-0
RMA History :00

WIC Slot 1:

T1 (1 port) WAN daughter card WAN daughter card
Hardware revision :1.0
Board revision :B0
Serial number :0022743844
Part number :800-03568-02
Test history :0x00
RMA number :00-00-00
Connector type :PCI
EEPROM format version 1
EEPROM contents (hex):
0x20: 01 20 01 00 01 5B 0B 24 50 0D F0 02 00 00 00 00
0x30: 58 00 00 00 00 12 05 00 FF FF FF FF FF FF FF

*!--- Note: If a PVDM is present in the PVDM slot but is not shown
!--- in show diag, try to reseat the PVDM module.*

The example output above shows the following:

- Slot 0 ASI 81 8–port FXS
- Slot 1;MFT– T1

Using the show voice dsp Command

The **show voice dsp** command shows how many DSPs are Available, Reserved, or Free.

```
ASI81# show voice dsp

!--- 8 analog channels are available (corresponds to the
!--- number of analog VIC ports, since the ASI 81 contains
!--- an 8-port FXScard).

Current total analog signaling channels: 8

!--- 30 digital channels available.  Calculated by
!--- multiplying 6 Chan/DSP x 5 DSP (loaded with High
!--- Complexity Digital image).

Current max allowed digital timeslot for voice: 30

!--- Information for transcoding number of Transcoding
!--- channels in use.

Current active transcoding sessions: 0

!--- Total reserved (allocated) channels for transcoding
!--- that are not active (in use).

Current free transcoding sessions: 2

!--- Free DSP resources that can be allocated for transcoding
!--- (can be added to the free transcoding field).

Current allocable transcoding sessions: 4

!--- There are two DSP groups (see the Grouping DSP Resources section
!--- of this document).

Current number of DSP group: 2

Group 0:
  Current allocated analog signaling channels: 8
  Current free analog signaling channels: 0
  Current allocated digital signaling channels: 0
  Current free digital signaling channels: 0

!--- VIC ports that are served by this DSP group (all
!--- analog VIC ports).

  Port(s) served: 0/0  0/1  0/2  0/3  0/4  0/5  0/6  0/7
  Current Available MIPS: 300

!--- DSPs that are not reserved for any application.
!--- Current Unused DSP: 0/3.
!--- (The following table gives detailed information on
!--- each DSP that is reserved or used, but might not
!--- necessarily have active calls on it. Active calls
!--- are illustrated by the Voice/Xcode Chan field.)

SPMM  DSPRM  State  Image  D-sig    D-sig  A-sig    A-SIG  Mips  Voice/Xcode
Dsp   Dsp                allocate free   allocate free   Free  Chan
```

```

0/0  0      UP      FIXMC  0      0      4      0      100  0
0/1  1      UP      FIXMC  0      0      4      0      100  0
0/2  2      UP      XCODE  0      0      0      0      100  0

```

Group 1:

```

Current allocated analog signaling channels: 0
Current free analog signaling channels: 0
Current allocated digital signaling channels: 24
Current free digital signaling channels: 6
Port(s) served: 1/0
Current Available MIPS: 500
Current Unused DSP: None

```

SPMM	DSPRM	State	Image	D-sig	D-sig	A-sig	A-SIG	Mips	Voice/Xcode
Dsp	Dsp			allocate	free	allocate	free	Free	Chan
1/0	0	UP	FLEX6	6	0	0	0	100	0
1/1	1	UP	FLEX6	6	0	0	0	100	0
1/2	2	UP	FLEX6	6	0	0	0	100	0
1/3	3	UP	FLEX6	6	0	0	0	100	0
1/4	4	UP	FLEX6	0	6	0	0	100	0

The example output above shows the following total DSPs:

- FXS: 2 DSPs; for Transcoding (provides two transcoding channels) and 1 DSP not used.
- PVDM-16: 4 DSPs (DSP Group 0) Serves 8-port FXS module, medium complexity.
- PVDM-20: 5 DSPs (DSP Group 1) 5 DSPs available for T1, 24 signaling channels are open, no active calls.

Grouping DSP Resources

DSP resources are grouped in MRP or ASI cards to better serve different configurations. There can be either a single DSP group or two DSP groups. Grouping, if needed, always happens on the boundary of the PVDM. In order to have two DSP groups, two PVDMs must be present.

The following are examples of DSP grouping:

- Mixed analog VIC and T1 or E1 VWIC requires two DSP groups. For example, slot 0 has a VIC-2FXS and slot 1 has a 2MFT-T1.
- Mixed BRI VIC and T1 or E1 VWIC requires two DSP groups. For example, slot 0 has a VIC-2BRI NT/TE and slot 1 has a 1MFT-E1
- Only T1 or E1 VWIC(s) are present, but two **tdm clock** commands have been entered. Defining two clock sources requires two DSP groups.
- If only an analog VIC or a BRI VIC are present, there is one DSP group. All the VIC ports on the MRP or ASI card can share a single PVDM.
- ASI 160 always has one DSP group because it is considered to have only one analog VIC with 16 ports.
- Transcoding can use DSPs in any DSP group. It is not restricted by the DSP group boundaries.
- Due to a limited number of timeslots on the TDM bus on ASI and MRP cards, up to 8 DSPs are usable if there is a single DSP group (4 DSPs from each PVDM). As a result, even if two PVDM-20s are installed on the MRP or ASI card and both PVDM-20s appear in the **show diag** command output only 8 DSPs are usable (which can be seen by entering **show voice dsp**). Therefore, in this case, a PVDM-20 is treated as a PVDM-16. In a two-DSP-group scenario, however, all DSPs on the PVDM-20 are available.
- Each DSP group serving analog ports needs one DSP for each VIC. Therefore, two analog VICs need two DSPs.

Example Calculation How many DSP modules are required for an ASI81 card with an 8-port FXS fixed in Slot 0 and a 1MFT-T1 in Slot 1?

	# DSPs for DSP Image	×	Channels/DSP for the Codec	=	Total Channels	PVDM Required
FXS Port	Using a Medium Complexity Analog DSP Image					
Codec G.711 or G.729a	2	×	4	=	8	PVDM 8
T1 Trunk	Using a High Complexity Digital DSP Image and assuming 80% of calls will be G.711 and 20% will be G.729a					
Codec G.711	3	×	6	=	18	
Codec G.729a	2	×	3	=	6	
(Total Channels for both Codecs)	(18 + 6 = 24)					PVDM 20

Note: The PVDM-8 for slot 0 can be increased to PVDM-20 to support 6 transcoding channels (2 channels of transcoding for each DSP × 3 DSPs = 6 channels).

Transcoding

Transcoding is needed when the calling and called parties can not use the same codec type. Codec incompatibility can also be caused by a failure when negotiating a common codec. For example, it is possible to have two Cisco voice gateways (such as an MRP or ASI) use different codecs and require transcoding to communicate.

- Any MRP or ASI card with DSP can be configured to act as transcoder. Transcoding can use any DSPs in an MRP or ASI card, regardless of how it is grouped.
- The reservation of DSP resources for transcoding is done by entering Cisco IOS commands, and a DSP that has been reserved can be released through Cisco IOS as long as transcoding is not using the DSP. Any changes made to the DSP partitioning take effect without reloading the MRP or ASI card.
- The MRP or ASI card uses the Skinny Client Control Protocol (SCCP) stack and can support VIC requirements and transcoding at the same time.
- The number of DSPs that can be reserved for transcoding is user-configurable.

Choosing the DSP Firmware Image

The DSP firmware image is downloaded to the DSP when it is reserved for transcoding. The command **show voice dsp** will show that the **Image** field has **XCODE** for such a situation.

Currently, the DSP firmware supports transcoding between the following pairs of codecs:

- G.723.1 and G.711u-Law
- G.723.1 and G.711a-Law
- G.729 and G.711u-Law

- G.729 and G.711a–Law
- G.711u–Law and G.711a–Law

Transcoding between low-bit-rate codecs such as between G.723.1 and G.729 is not supported, but it can be achieved by using two transcoding sessions with G.711u as a bridge.

Determining How Many DSP are Needed for Transcoding

DSP resources must be reserved before the MRP or ASI card can act as a transcoder. MRP or ASI cards are flexible about how many transcoding channels to reserve. A DSP is required for every two transcoding channels. Each DSP supports two full-duplex transcoding channels.

Understanding How DSP Resources are Allocated

When the MRP or ASI card boots, DSP resources are statically allocated for analog VICs and BRI VICs first. These DSP resource allocations cannot be changed. In the **show voice dsp** command output, these DSPs are represented with a value of **FIXMC** or **FIXHC** in the **Image** field, depending on whether high or medium complexity DSP firmware is being used. The remaining DSP resources can be allocated to T1/E1 VWICs and transcoding, as needed.

For T1/E1 VWICs, DSPs are reserved by defining a ds0-group or pri-group under the individual T1 or E1 controller. A DSP is reserved if it hosts a signaling channel for the T1/E1 VWIC. Such a reserved DSP has a non-zero value in the **D-sig Allocate** field in the **show voice dsp** command output.

Each DSP supports two full-duplex transcoding channels. The MRP or ASI card can be directed to set aside all unreserved DSP resources for transcoding by entering the **sccp transcode** command without the **channel** option. By default, Cisco IOS reserves the minimum number of DSP resources based on the **channel** option, if present.

Once a DSP is reserved for a particular purpose, its allocation can not be changed without entering Cisco IOS commands. This means that DSPs reserved for transcoding cannot be used by a T1/E1 VWIC even if there is no active transcoding session on the DSP and vice versa.

The guidelines for transcoding are as follows:

- Transcoding shares the same DSP pool with other analog and digital VICs.
- MRP resources can be shared only for transcoding. If no VICs are present, the MRP can use up to eight DSP resources for transcoding.
- SW transcoding is a by-pass in Cisco IOS. Cisco IOS uses SW transcoding when the in-and-out codecs are the same.
- Currently, the MRP provides TDM-to-IP transcoding which enables compressed WAN calls to be decompressed and sent to the voice interface (for example, to the G.711 Public Switched Telephone Network [PSTN]).

Configuring Transcoding On MRPs and ASIs

The following IOS commands are used in global configuration mode to configure transcoding:

- **sccp local interface** *IP address [port]*

The command above defines which interface to use for registration with Cisco CallManager and for packet transmission and reception. For MRP or ASI cards, it should typically be FastEthernet 0/0.

- **sccp manager** *IP address1 [port] [priority1]*

sccp manager *IP address2 [port] [priority2]*

sccp manager *IP address3 [port] [priority3]*

sccp manager *IP address4 [port] [priority4]*

The command above registers the MRP or ASI card with Cisco CallManager as a transcoder. Four Cisco CallManager instances may be entered, each with a different priority. The *priority* options (values ranging from 1 [highest] to 10 [lowest]) specify the priority of each Cisco CallManager instance.

- **sccp transcode** *[ip-precedence value] [channels number]*

The command above reserves transcoding channels. If the *channels number* option is omitted, the MRP or ASI card automatically reserves all of the unused DSP resources for transcoding and informs the user of the number of available transcoding channels by sending a syslog message.

Hardware Conferencing

Hardware conferencing uses a DSP to act as a conference bridge. Before entering the conferencing DSP, voice streams are normally converted to G.711 through a transcoding DSP if the incoming codec is not G.711 (mixed codec conferencing). Therefore, a DSP is not necessarily required for conferencing.

In the Cisco ICS 7750, mixed codec conferencing is accomplished by using the MRP or ASI card as the transcoder and Cisco CallManager as the conference bridge.

Related Information

- [Cisco ICS 7750 Getting Started Guide](#)
- [Installing Memory and PVDM Modules in ASI Cards, MRP Cards, and SPE Cards in the Cisco ICS 7750](#)
- [Release Notes for the Cisco ICS 7750 for Cisco IOS Releases](#)
- [Field Notices](#)
- [Voice Software](#)
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Updated: Mar 30, 2006

Document ID: 7123
