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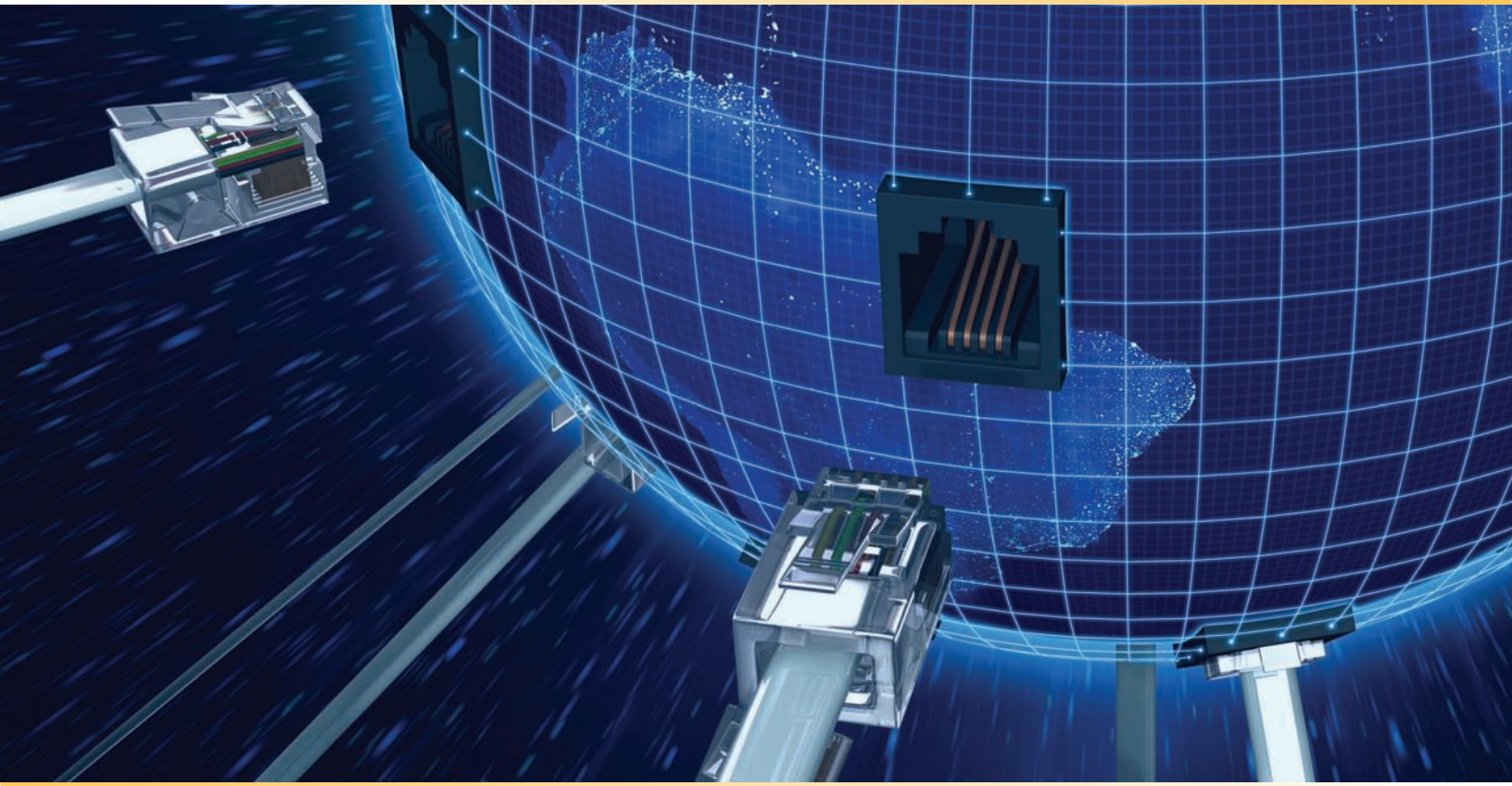
Toshiba Strata® CIX™ Systems

Technical Application Notes



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About Toshiba Strata® CIX™ and Broadvox SIP Trunking

Session Initiation Protocol (SIP) is an application layer protocol used for establishing sessions in an Internet Protocol (IP) network. Broadvox SIP Trunking allows the Toshiba Strata® CIX™ system to get Primary Rate Interface (PRI)-like services from an Internet Telephony Service Provider (ITSP) using SIP. A SIP Trunk allows a Strata CIX system to connect internal voice and private data traffic to the outside public network (Public Switched Telephone Network and public data) via Internet Protocol (IP.) In the traditional world, a business would have some type of dedicated lines to route voice traffic to the outside PSTN. These lines could be analog Plain Old Telephone Systems (POTS), T1 or PRI. These circuits are considered a PSTN gateway because they make the connection between internal callers and the outside callers.

With SIP Trunks, voice and connections to the outside world use the same IP WAN. When a user dials a call to send over the PSTN, the call routing goes over the WAN to the Internet Telephony Service Provider (ITSP) that is providing the SIP trunk. This ITSP will provide a connection to the PSTN through their equipment. The call goes from the CIX system to the SIP provider, who will act as a proxy, and send the call to the dialed destination. For incoming calls, the SIP trunk acts somewhat like a DID trunk; the dialed number is sent to the SIP provider and then routed over the IP Network to the CIX system. This routing is based on the URI and associated IP address.

Toshiba's SIP Trunk capabilities allow the Strata CIX system to communicate with a service provider natively over an IP circuit, which can be used to carry voice and data simultaneously. Inside the CIX system, voice is converted to data and sent to the service provider along the same circuit as the other data packets. This allows voice and data to use one circuit; it also allows data to use all of the bandwidth when no voice is present. Quality of Service (QoS) is managed by the service provider, allowing voice to instantaneously take priority over data.

SIP trunks are set up using the MIPU/GIPU card. The MIPU/GIPU is designed as a VoIP card, able to support IP stations, Strata Net IP and SIP trunks.

SIP trunks offer ISDN-like features over a data connection (i.e. a T1 circuit). However, unlike a traditional T1 circuit, a SIP trunk enabled circuit does not have to be physically provisioned and divided to separate the voice channels from the data channels.

Requirements

- Strata CIX Hardware: CIX40, CIX100, CIX200, CIX670 or CIX1200
- Strata CIX Software: R5.20 MT14 or higher
- MIPU/GIPU, MIPU16/MIPU24 (MIPU01.18 or higher)/GIPU8
- Network eManager®: V5.10 A18 or higher
- Service provider: Providers are listed in this document, but contact the Toshiba Sales Applications Desk for the latest list.
- License LIC-CIX-SIPT-CH
- Broadvox Provided **Version N** SIP Trunk

Purpose, Scope and Audience

This technical application note describes the configuration of Toshiba for the Broadvox GO! SIP Trunking service. This document has a technical audience in mind – specifically IT professionals with some experience in PBX administration and familiarity with VoIP technologies. This document is not for business administrators or people in other non-technical careers. In order to successfully use this document to deploy Broadvox GO! SIP Trunking service, you will need to possess the following skills, or have access to professionals or consultants with the following skills:

- Familiarity with PBX systems, including:
  - Trunk configuration
  - Calling plan configuration
  - Extension configuration
  - Mailbox configuration
- Familiarity with Session Initiation Protocol (SIP)
- A complete understanding of your internal network structure, Network Address Translation (NAT) setup, and firewall setup
- A complete understanding of your public Internet connectivity

Broadvox can only provide support for Toshiba to the extent covered in this Technical Application Note and the included reference configuration, so if your level of technical expertise does not include the above skills, it is recommended that you obtain the services of a software professional.

What is SIP Trunking?

Before you begin deploying SIP Trunking with Toshiba gateways, it is important to understand the basic definition of SIP Trunking.

Session Initiation Protocol (SIP) is the common signaling standard for real-time communications including Voice over Internet Protocol (VoIP). SIP is an open standard which allows carrier voice equipment to interoperate seamlessly with customer premise equipment (CPE). The logical voice channel established between them is referred to as a SIP Trunk; a virtual phone line that utilizes a broadband connection for access.

Business Advantages of SIP Trunking

Businesses have installed TDM phone systems (PBX or Key Systems) that work just fine and the owners and managers of the businesses see no need to replace them just yet. However, the owners and managers of these businesses would like to take advantage of some of the benefits of IP Telephony, perhaps by saving money on the monthly service via SIP Trunking, taking advantage of a Unified Communications Solution, or utilizing VoIP services like Extended Local Numbers. Until now, there has not been a cost-effective way for these businesses to phase IP telephony into their existing infrastructure.

Broadvox and Toshiba have created a bundle that solves this problem with a seamless, cost-effective solution. By utilizing a Toshiba Strata CIX system along with Broadvox SIP Trunking, businesses can gain the cost savings that VoIP provides along with the features and functions of IP Telephony without introducing migration problems. There is no need to dispose of any existing equipment, no need for extensive end-user training, and access to a very nice phased implementation takes place instead of a “forklift upgrade.” Best of all, the monthly savings on SIP Trunks combined with the innovative purchase arrangements make this solution extremely affordable.

Technical Challenges

The biggest technical challenge facing organizations who want to move to SIP Trunking while retaining their existing TDM PBXs is interoperability. The SIP trunks must be converted to a TDM format that is compatible with that specific PBX, allowing the TDM circuits from the PBX to be moved from the PSTN trunks to the SIP trunks with little or no adjustments to the PBX.

Business Challenges

A key business challenge for both the enterprise and reseller is separating the sale of SIP Trunking from any potential upgrades or replacement of the existing TDM PBX and avoiding the protracted evaluation process required by the customer to select a new PBX, work out a migration plan and then execute a cut over.

In addition, interoperability between the SIP Trunking and the existing TDM PBX is important, avoiding any expensive upgrades or modifications to the PBX. If the PBX needs new cards installed, if the software requires upgrades or significant administration time is required, the cost saving of the SIP trunks could be overshadowed by upgrade and installation costs.

Introduction to Broadvox

Choosing the SIP Trunk Version

Broadvox offers two versions of SIP Trunking to meet customer requirements. By default, Broadvox places most new customers on Version N. Version N supports dynamic IP addresses using registration, offers NAT Traversal and offers automatic fall-back forwarding to PSTN numbers, either on a per-trunk basis, or a per-number basis.

If using more than 1,000 simultaneous calls is anticipated, consider using Version S. The Version S trunks do not support registration, NAT Traversal or automatic fall-back forwarding to PSTN numbers. However, Broadvox can offer up to four separate gateways across three cities to which calls may be sent and from which calls may be received (by default, connectivity to two is offered, but for very large customer implementations, additional gateways can be added). In addition, the Version S gateways are rated for a much higher call capacity than the Version N gateways, and offer more granular control of the specific signaling and media profiles for the trunk. The more granular control allows more flexibility in accommodating any special requirements.

Sending Calls to Broadvox

IP addresses to which calls may be sent and from which calls may be received are listed in the Welcome letter from Broadvox. The Toshiba Strata CIX System supports IP addresses and DNS records. Broadvox uses two Welcome letters depending on which version of services are deployed. Version S uses two entirely separate servers to provide high availability. Version N uses a cluster and a virtual IP address to provide high availability. In the figure below, the three main sections for Version S are listed.

Allow These Media IPs - UDP Ports Up To 65535			
64.158.162.71	64.158.162.100	64.152.60.71	64.152.60.164

Trunk Number	Turn-up Ticket	Trunk Type	BTN	Source IP
1234567	23456	GO!Anywhere	5551234567	127.0.0.1
2345678	23456	GO!Domestic	NA	127.0.0.1

Trunk Number	DNS A Record	DNS SRV Record	IP Addr 1	IP Addr 2
2345678	east.gd.broadvox.net	psrveast.gd.broadvox.net	64.158.162.76	64.152.60.76
1234567	east.ga.broadvox.net	psrveast.ga.broadvox.net	64.158.162.75	64.152.60.75
	east1.ga.broadvox.net	psrveast1.ga.broadvox.net	64.158.162.101	64.152.60.165
	east2.ga.broadvox.net	psrveast2.ga.broadvox.net	64.158.162.102	64.152.60.166
	east3.ga.broadvox.net	psrveast3.ga.broadvox.net	64.158.162.103	64.152.60.167
	east4.ga.broadvox.net	psrveast4.ga.broadvox.net	64.158.162.104	64.152.60.168

Figure 1 - Three Main Sections from Version S Welcome Letter

The first section lists the IP addresses from which media will be received. The firewall must allow these IP addresses. Media may be received from any User Datagram Protocol (UDP) port from 1024 to 65535 from any of the listed addresses.

The second section lists each of the trunks, the service ticket number containing the turn-up information, the type of the trunk, the BTN for the trunk, if any and the

source IP address from which Broadvox expects to receive calls and to which calls are to be directed. The Source IP listed **must** match the External IP Address to be used.

The third section lists signaling addresses for each trunk group. Note that the trunk group number listed matches the trunk group number from the second section. This section tells which trunk sends to which IP address. If the incorrect IP address for the trunk is input, the charge for calls may be more than anticipated.

For example, if the IP addresses are reversed and GO!Domestic traffic is sent to the .75 address listed for the GO!Anywhere trunk while the GO!Anywhere traffic is sent to the .76 address listed for the GO!Domestic trunk, then GO!Domestic calls will be rated as long distance from the GO!Anywhere trunk LD minutes, if LD minutes have been purchased. If a bundle of LD minutes on the GO!Anywhere trunk have not been purchased, there is typically a charge of 3.5 cents per minute and GO!Anywhere calls are rated as long distance calls from the LD bundle purchased for the GO!Domestic trunk. This could result in much higher fees than anticipated. Verify that traffic is sent to the correct address. If different IP addresses are used for each trunk, then the calls will fail if sent to the incorrect location.

In the Version N Welcome letter, the media IP addresses are different and there is only one signaling IP address for each trunk instead of two, as shown in Figure 2 below. Also, note that the Version N Welcome letter contains a Password field for each trunk. This field will contain a password for the trunk if using registration. If registration is not used, the Source IP field will be filled out instead. When using registration, the IP address is obtained from the register request.

Trunk Number	Turn-up Ticket	Trunk Type	BTN	Source IP	Password

Trunk Number	DNS A Record	DNS SRV Record	IP Addr 1
	dfwnx01gd.pa.broadvox.net	dfwnx01gd.psrv.broadvox.net	209.249.3.57
	dfwnx01ga1.pa.broadvox.net	dfwnx01ga1.psrv.broadvox.net	209.249.3.59
	dfwnx01ga2.pa.broadvox.net	dfwnx01ga2.psrv.broadvox.net	209.249.3.100
	dfwnx01ga3.pa.broadvox.net	dfwnx01ga3.psrv.broadvox.net	209.249.3.101
	dfwnx01ga4.pa.broadvox.net	dfwnx01ga4.psrv.broadvox.net	209.249.3.102

Figure 2 - Version N Welcome Letter IP Addresses

SRV Records

Service records (SRV) are a form of Domain Name System (DNS) record. They contain information about where to send requests for a particular service offered at a specific domain. In the case of Broadvox GO! SIP Trunking, they provide the IP addresses, port numbers, and preferences to use for sending SIP calls over UDP to Broadvox. The SRV location to use for sending calls to Broadvox for each trunk group is included in the Welcome letter.

Choose the correct server based on the information in the Welcome letter. There is one caveat with using DNS records. The DNS query is cached and may use only one IP address for signaling to/from a peer. To configure servers to provide proper redundancy, Broadvox recommends using IP addresses instead of DNS. If DNS records are used instead of IP addresses and one of the servers is removed from DNS to perform a scheduled maintenance, the server will continue to try to target that IP. This will either add a small amount of post dial delay to calls, or cause calls to start failing until DNS is re-checked. Post dial delay (PDD) is the period of time between dialing a number finishes and ringing starts. Typically, the added delay will be less than 3 seconds but usually will be under 1 second.

Testing SRV Records

Most Broadvox GO! SIP Trunking customers like to ensure the DNS entries are functioning or they like to look up the actual IP addresses, however, performing a standard DNS query on the SRV records will fail. In a Windows environment, perform the query using the **nslookup** command at a command prompt, as shown in Figure 3.

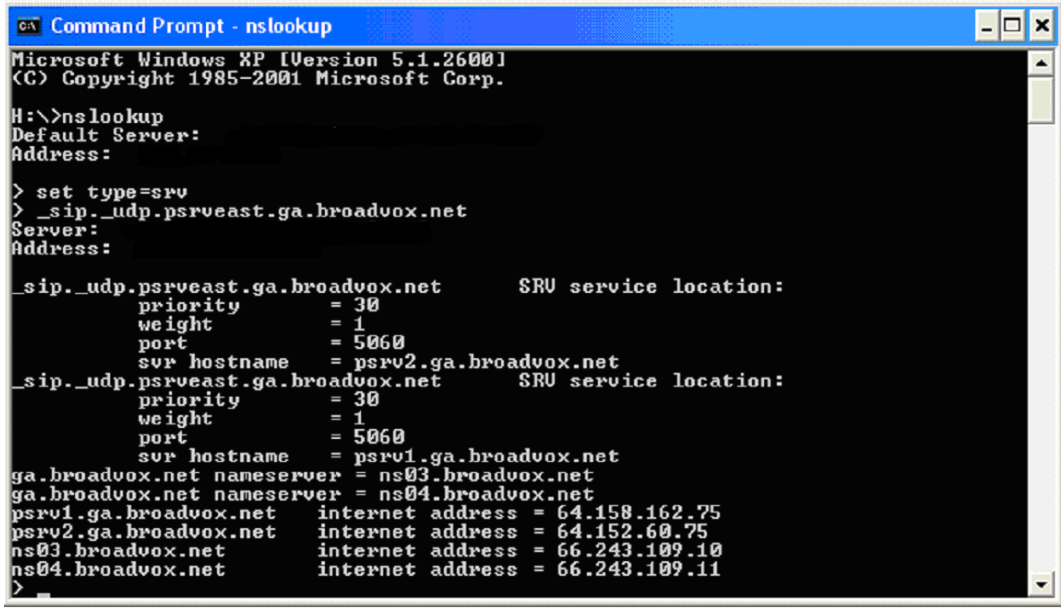

Figure 3 - SRV Lookup in Windows XP

As shown above, a SRV record consists of a service type definition (_sip), a transport definition (_udp), and the domain (psrveast.ga.broadvox.net). A PBX that supports SRV records will automatically add the service and transport definitions as a prefix to the domain before performing the query. The query returns a priority, weight, port and hostname for each entry. The query also returns the “A record” entries for each hostname, which provides the IP address for each host.

In a UNIX or Linux environment, a query on SRV records using the **`dig`** command can be performed:

```
$ dig SRV  sip. udp.psrveast.ga.broadvox.net

; <<>> DiG 9.3.3 <<>> SRV  sip. udp.psrveast.ga.broadvox.net
;; global options:  printcmd
;; Got answer:
;; ->>HEADER<<- opcode: QUERY, status: NOERROR, id: 16950
;; flags: qr aa rd ra; QUERY: 1, ANSWER: 2, AUTHORITY: 2, ADDITIONAL: 4

;; QUESTION SECTION:
; sip. udp.psrveast.ga.broadvox.net. IN SRV

;; ANSWER SECTION:
 sip. udp.psrveast.ga.broadvox.net. 600 IN SRV  30 1 5060 psrv2.ga.broadvox.net.
 sip. udp.psrveast.ga.broadvox.net. 600 IN SRV  30 1 5060 psrv1.ga.broadvox.net.

;; AUTHORITY SECTION:
ga.broadvox.net.        600     IN      NS      ns04.broadvox.net.
ga.broadvox.net.        600     IN      NS      ns03.broadvox.net.

;; ADDITIONAL SECTION:
psrv1.ga.broadvox.net.  600     IN      A       64.158.162.75
psrv2.ga.broadvox.net.  600     IN      A       64.152.60.75
ns03.broadvox.net.      7200    IN      A       66.243.109.10
ns04.broadvox.net.      7200    IN      A       66.243.109.11

;; Query time: 2 msec
;; SERVER: 66.243.109.10#53(66.243.109.10)
;; WHEN: Wed Nov  7 12:30:24 2007
;; MSG SIZE  rcvd: 236
```

SIP Trunking Configuration

SIP Trunking Configuration Example

The example shown below is a general system plan. Interoperability was confirmed using BroadvoxGO! **Version N** SIP Trunking. Please note that BroadvoxGO! **Version S** SIP Trunking is not compatible.

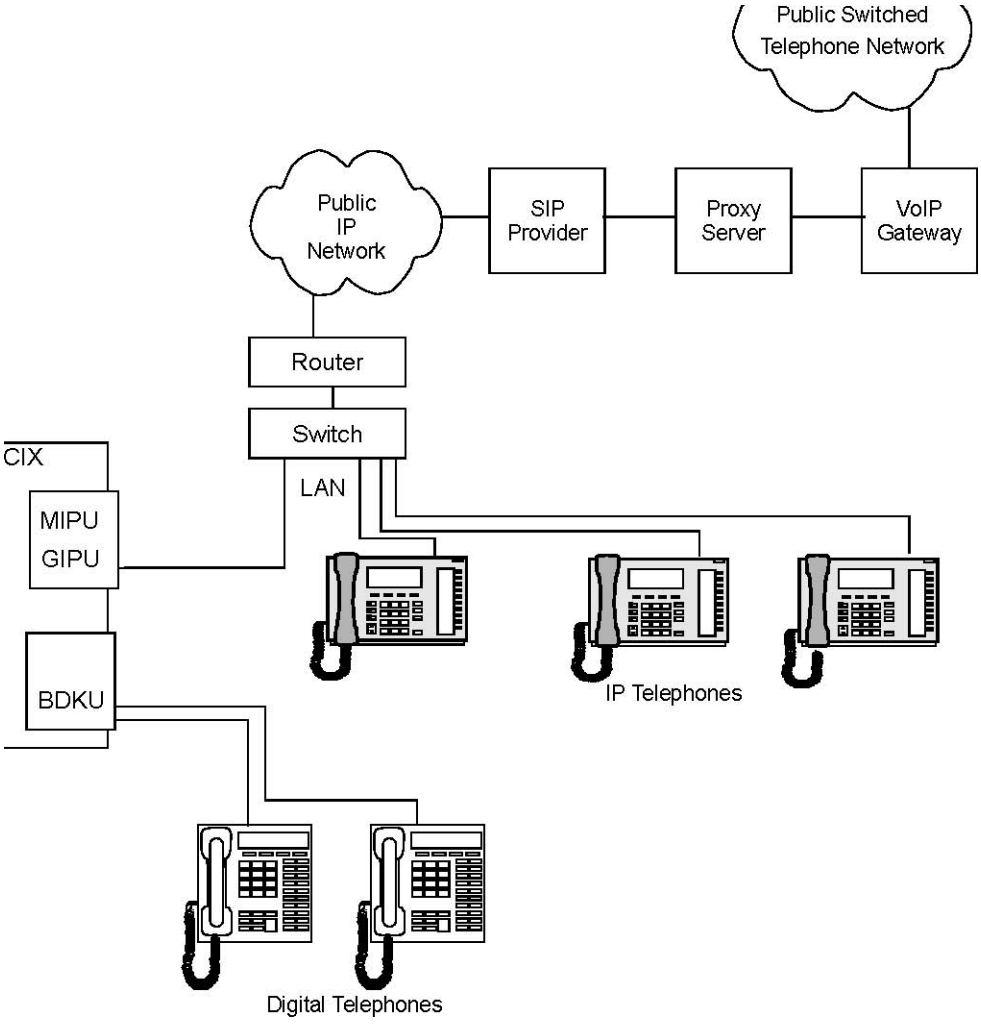

Figure 4 - CIX with SIP Trunking

There are many ways to set up a public IP address for the MIPU or GIPU. The most common way is to simply run an Ethernet cable from one of the WAN interfaces on the router straight to the MIPU/GIPU.

SIP Trunking Configuration Programming

The following pages show the general programming and configuration steps to implement a SIP trunk. Specific procedures for each provider are in the linked tables in this document.

Programming the MIPU/GIPU card

1. Set the card type for the slot holding the MIPU.
2. Choose the type of MIPU for use in the system. The number of channels entered must match the number of channels on the card.

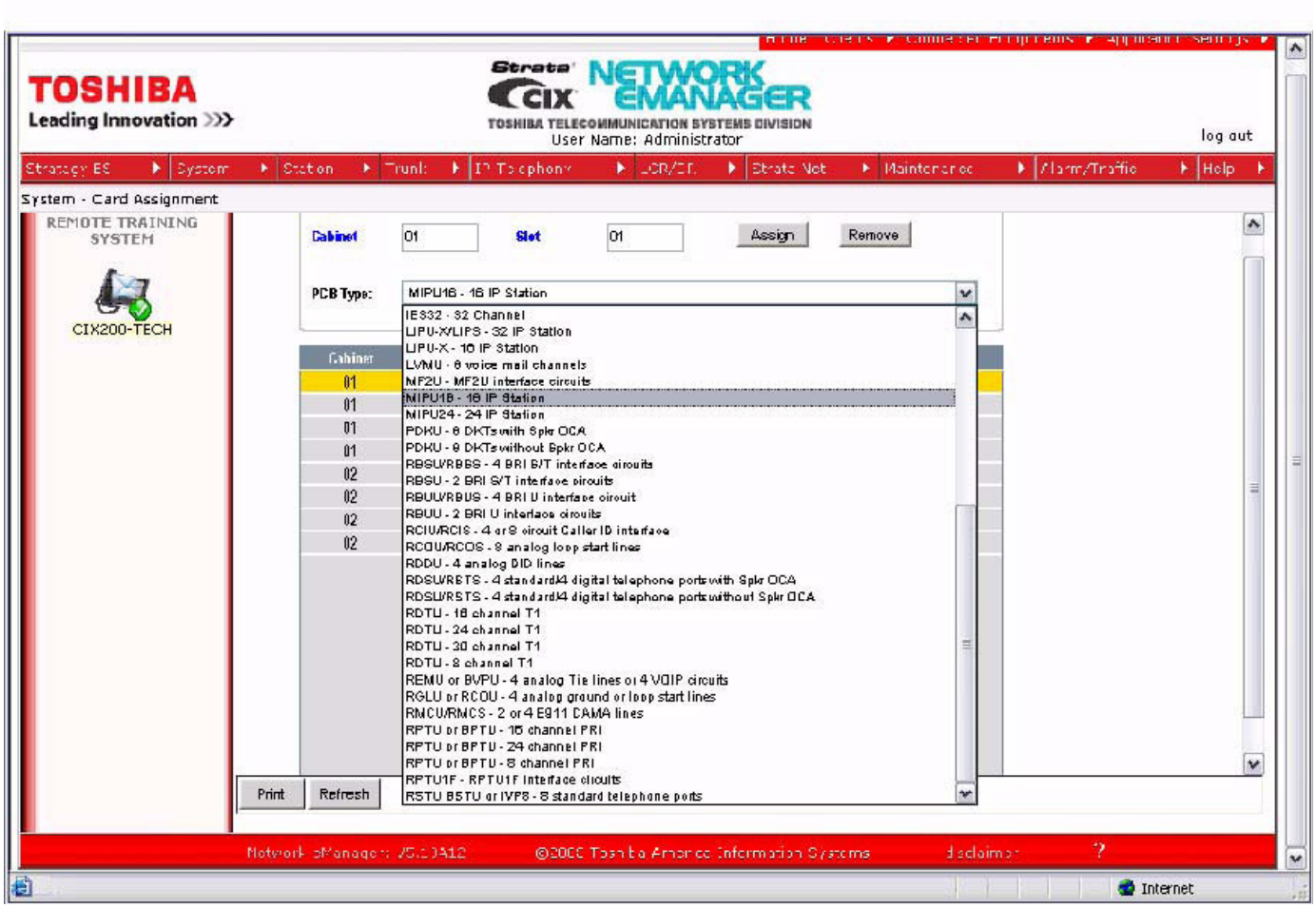

Figure 5 - Programming the MIPU/GIPU card

Programming the Incoming Line Group

1. For SIP trunk programming, start with an ILG and Outgoing Line Group (OLG) assignment.
2. For Incoming Line Group (ILG) programming, create an ILG group (in this example, group number 8).
3. FB01 - Trunk type: Select SIP.
4. FB03 - CO Service Type: DID.
5. FB11 - DID Digits: Number of incoming digits to use from the dialed number to route the call. Count the number of digits from the least significant digit.

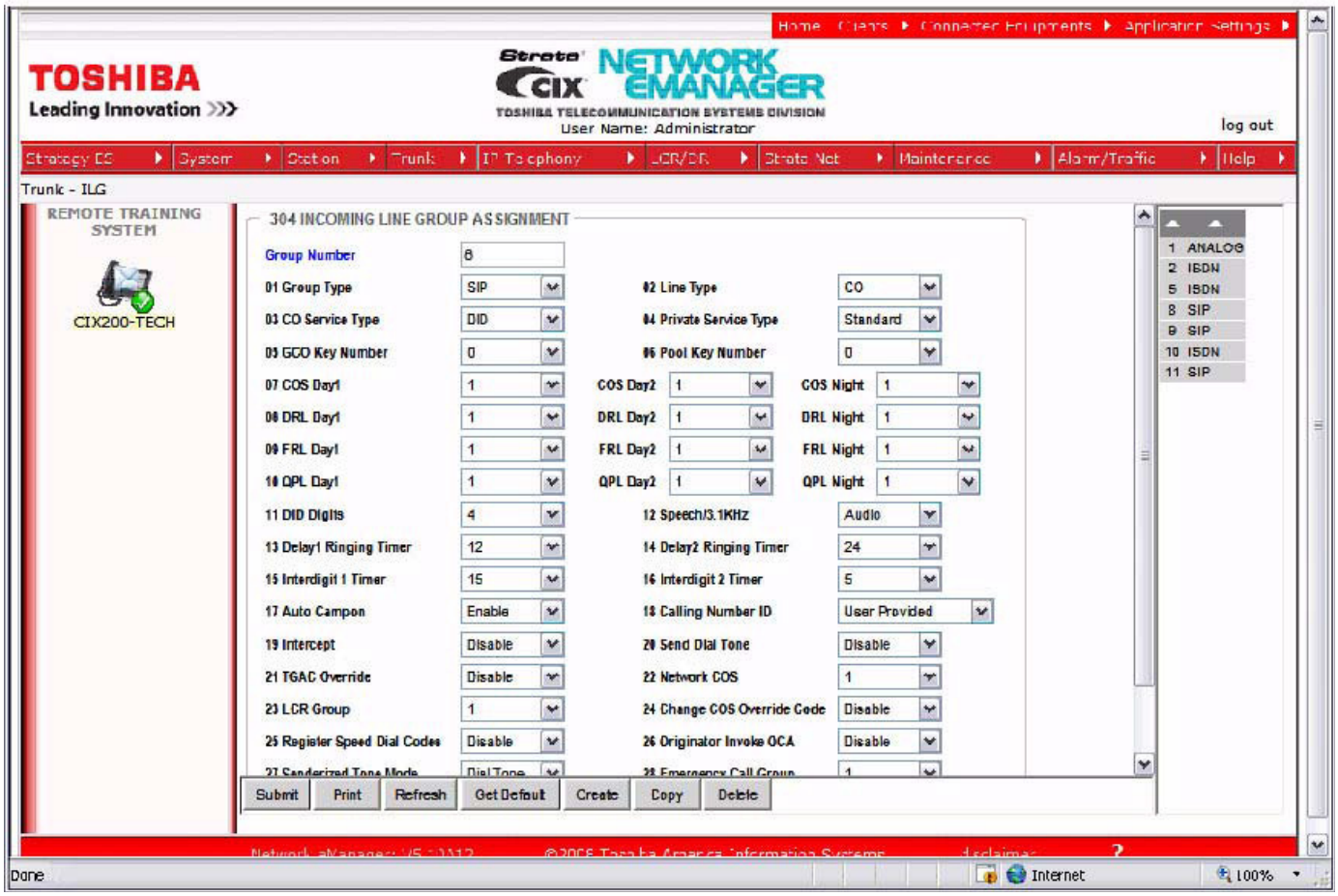

Figure 6 - Programming the Incoming Line Group

Note: SIP Trunking requires a license for each trunk. No channel group can successfully be programmed without a license.

SIP Trunking is supported by the MIPU/GIPU cards only. It can share connections of its ports between stations, IP telephones and other IPU devices. The IP telephones are the only devices that are flexibly

assignable. That is, SIP stations, VM ports, SIP trunks and attendant consoles are fixed resources. IP telephones do not need to be fixed assignments unless they are assigned a fixed equipment number (not 0000).

Assigning the Trunk DID Destination

1. FB01 - DID number: Incoming digits used to direct the dialed number to route the call. Count from the least significant digit side and limit to the number of digits set in Prog 204, FB11.
2. FB05-07 - Audio Day 1 Dst Type, Audio Day 1 Dst Digits, Day 2, Night: For example, Dialing Digits/Extension

Note: DID routing must be set up to route incoming SIP calls to their desired destination. This programming is the same as any other trunk group type. If the routing is not set up, incoming INVITES will fail instantly.

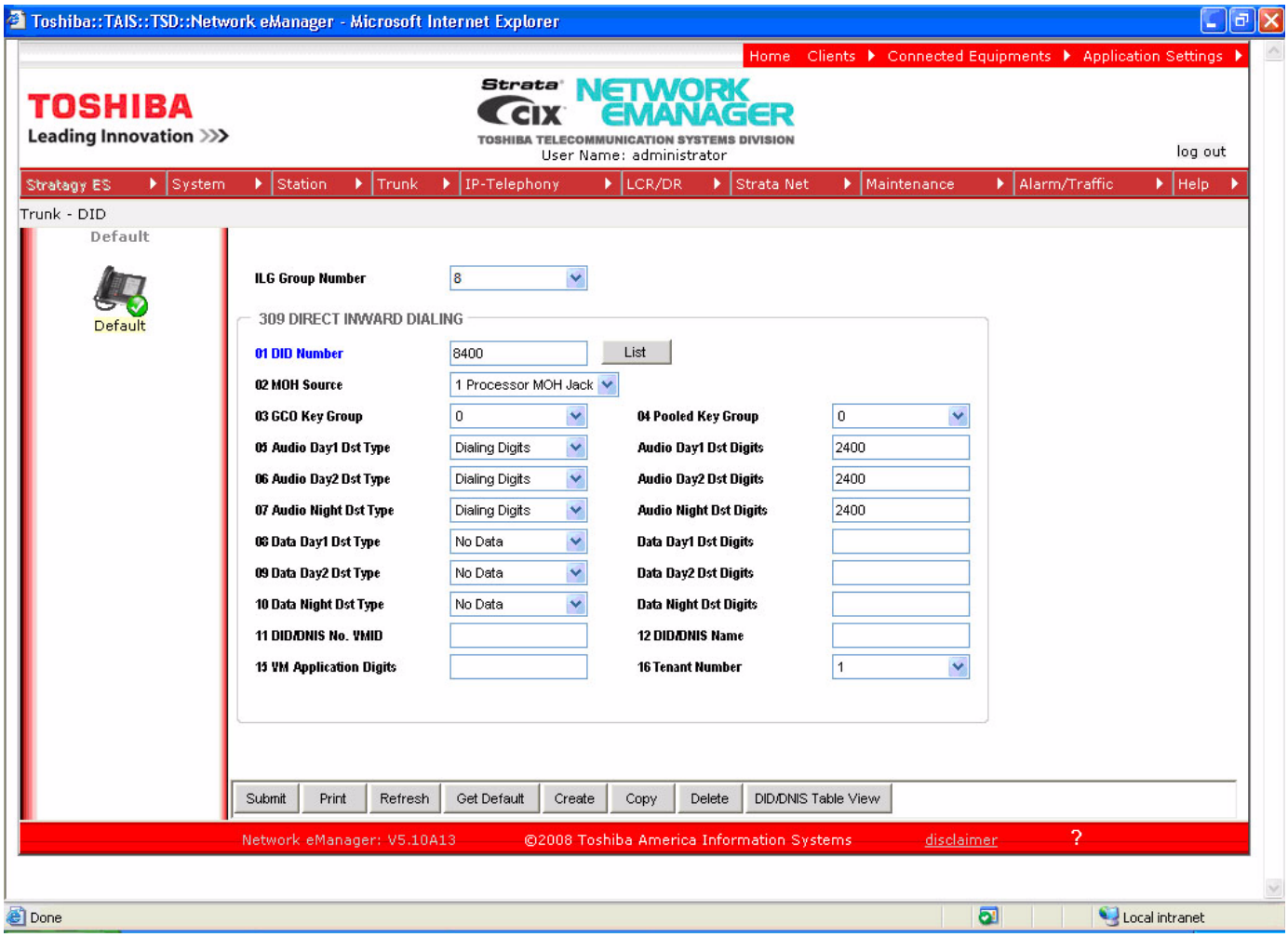

Figure 7 - Assigning the Trunk DID Destination

Programming the Outgoing Line Group

1. Create a new Outgoing Line Group (OLG) and designate it as SIP.
2. Set FB01 - Group type: SIP
3. The rest of the values are left at the default value.

Note: An OLG flexible access code must be created for this group.

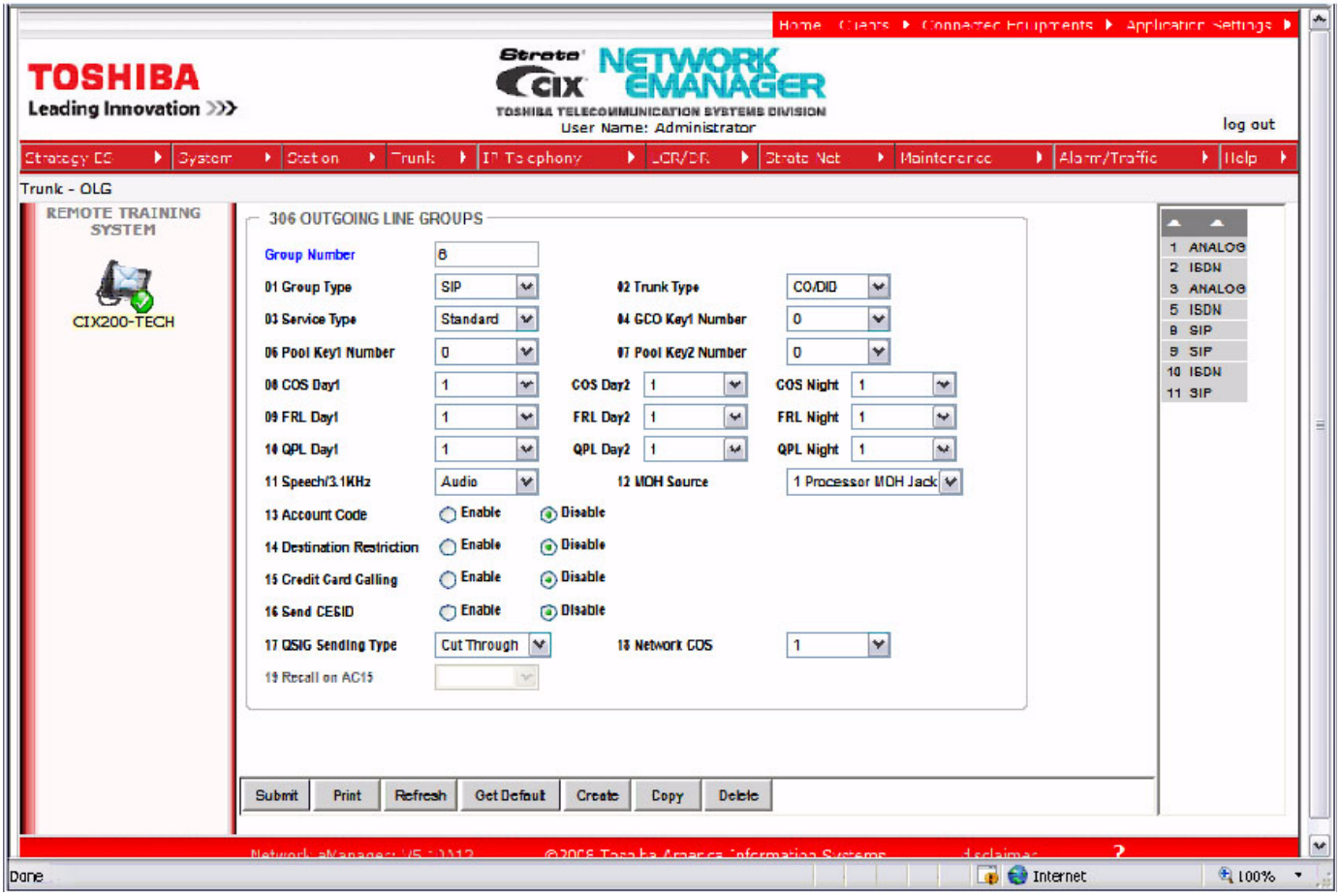

Figure 8 - Programming the Outgoing Line Group

OLG Flexible Access Code Programming

An access code is required for the OLG that was setup for the SIP Trunks. SIP Trunks can also be accessed using LCR.

1. FB00 - create an access code for the SIP Trunk (in this example, 8.)
2. FB01 - Feature name set to Line group access code; one access code for each OLG.
3. FB02 - Enter the OLG number set in Programming OLG on the previous page (in this example, 8.)

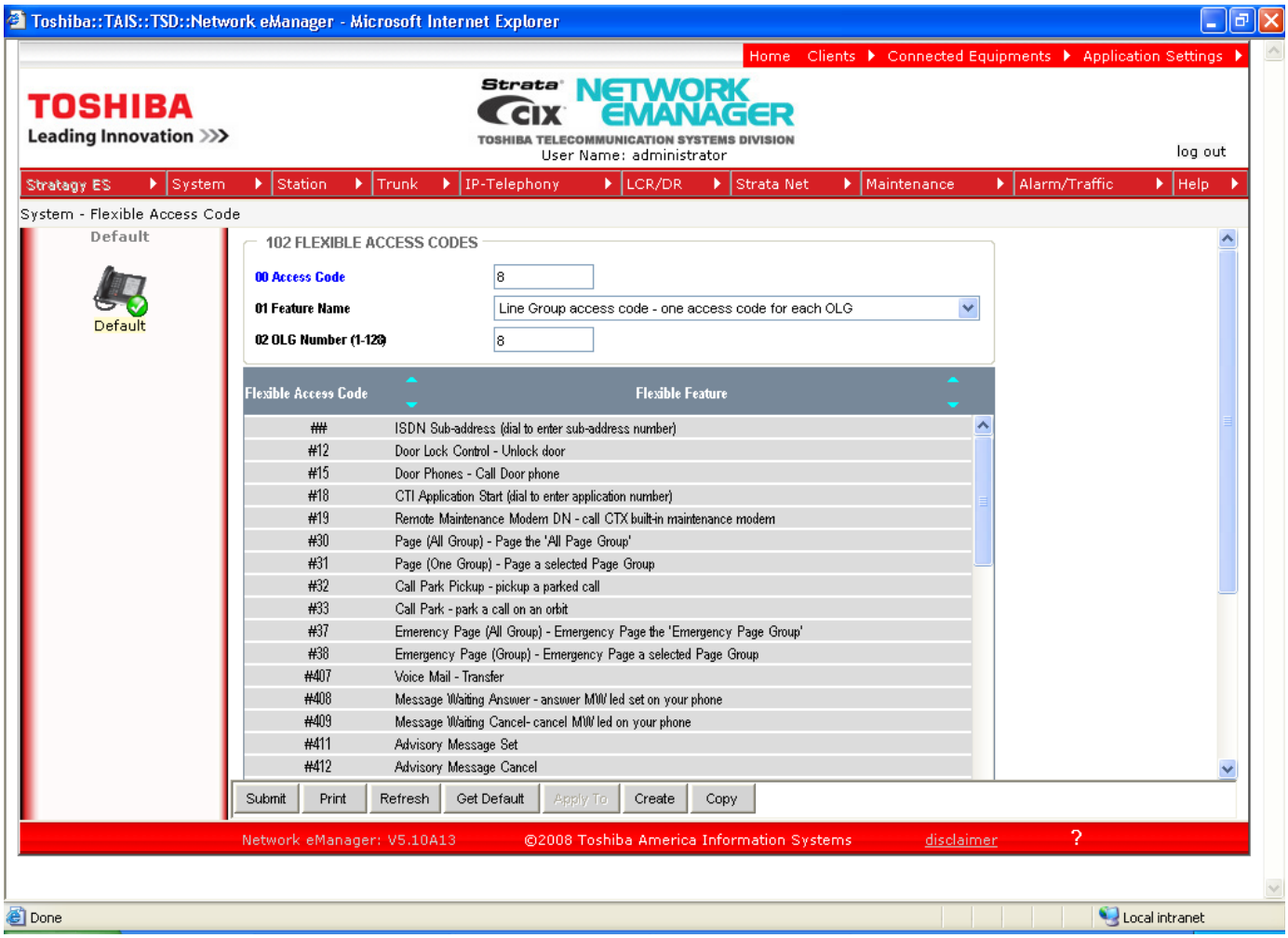

Figure 9 - OLG Flexible Access Code Programming

Creating the Channel Group

1. Go to IP Telephone > SIP Trunking. The first tab is Channel Group setting (Program 326.)
2. FB00 - SIP Trunk Channel Group: In this example SIP Trunk Channel Group 16 is created.
3. FB01 - Equipment Number: Enter the Cabinet number and slot number (four digits.) In this example, 0201.
4. FB02 - LAN interface number - 1 (MIPU/GIPU have one IP interface.)
5. FB03 - Strata Net Channels: Enter the TOTAL number of ports on this card that are going to be dedicated to SIP Trunking. On further tabs, each OLG or ILG can be for a different service provider and still be in the same channel group.
6. FB04 - RBT (Ring Back Tone) on incoming call: Enable the CIX to provide RBT.

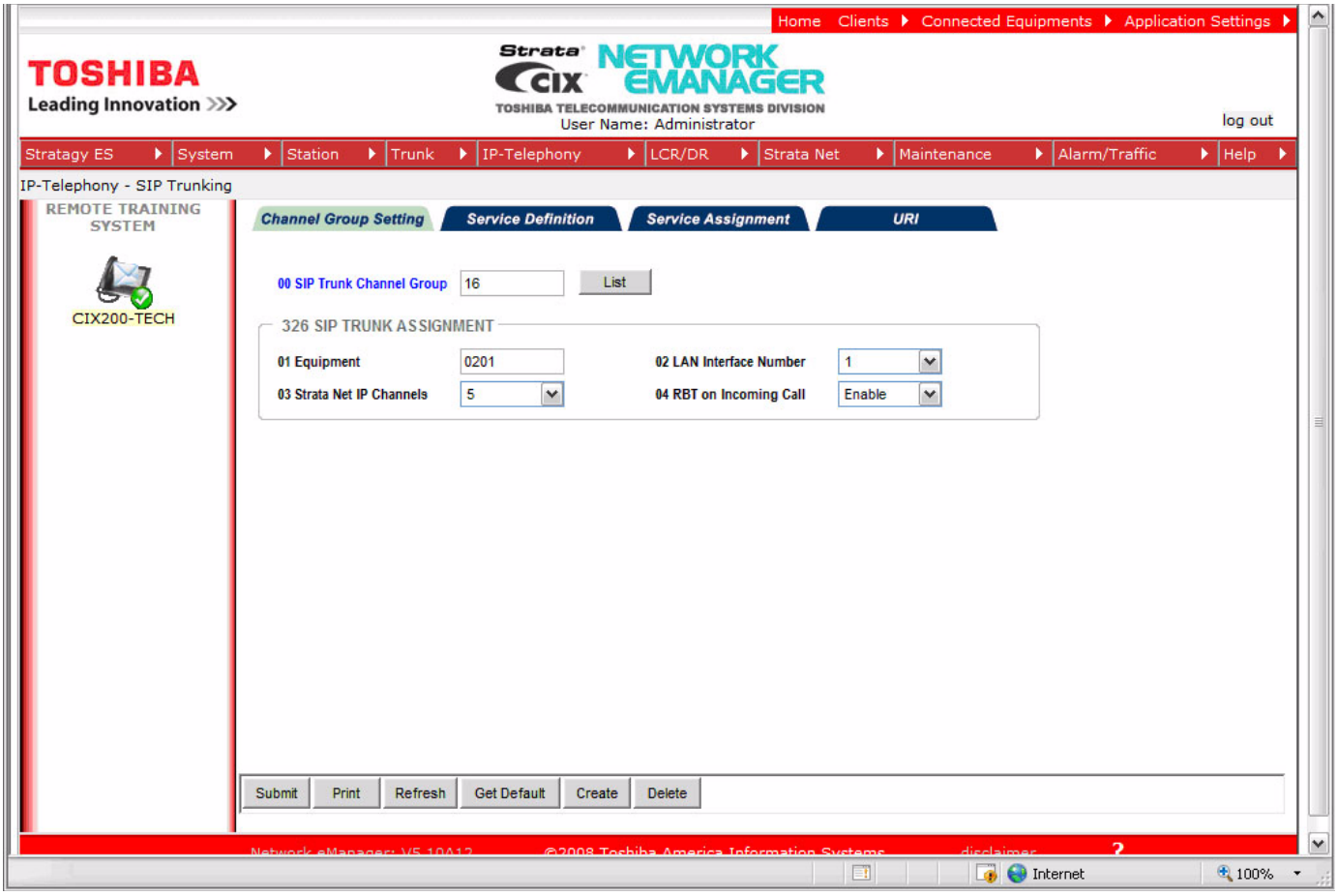

Figure 10 - Creating the Channel Group

Service Definition

The Service Definition tab is where the SIP Trunk provider is defined. There may be several SIP Trunk providers per Channel Group.

1. Create the Service Kind Table Index - Start with index 1 if desired (11 in this example.)
2. FB01 - Registration Mode: Broadvox Static NextPoint (Version N static) set to None: Broadvox Dynamic NextPoint (Version N Dynamic) set to Client.
3. FB02 - ILG: Use the previously created ILG number.
4. FB03 - OLG: Use the previously created OLG number.
5. FB04 - Effective Channel Number: Number of SIP Trunks from the SIP provider.
6. FB05 - Domain Name: Use the DNS SRV Record provided by Broadvox (i.e. dfwnso1ga1.psrv.broadvox.net). This name is specific to this installation.
7. FB06 - SIP Server: Enter the IP address (will be provided by Broadvox.)
8. FB12 - Disabled

Figure 11 - Service Definition

Note: The Toshiba Strata CIX is not certified with Broadvox for T.38 as Toshiba listed it as "not supported" during interoperability testing and did not test its functionality.

9. FB34 - SIP Trunk Options Interval: For Static NextPoint (Version N static) set to 60; for Dynamic NextPoint (Version N Dynamic) set to 0.
10. FB35 and FB37 - Set to IPU IP Address.
11. FB36 and FB38 - SIP Server IP Address

These are the only mandatory program entries. The rest can be left at the default values.

Figure 12 - Service Definition, continued

Service Assignment

This tab must be completed before entries can be made on the next tab (URI.) If not completed, entries attempted in the URI tab will not be saved.

Important Note: After URI entries are made in URI tab, changing this program will DELETE all the URIs programmed. Once this is set, do not change it.

1. FB01 - Channel Group (16 in this example.)
2. FB02 - Select the Service Number by clicking on one of the table's line entries. Use the dropdown box to select the SIP Trunk provider to which the URIs belong. This should match the Service Kind Table Index number of the desired SIP Trunk provider.

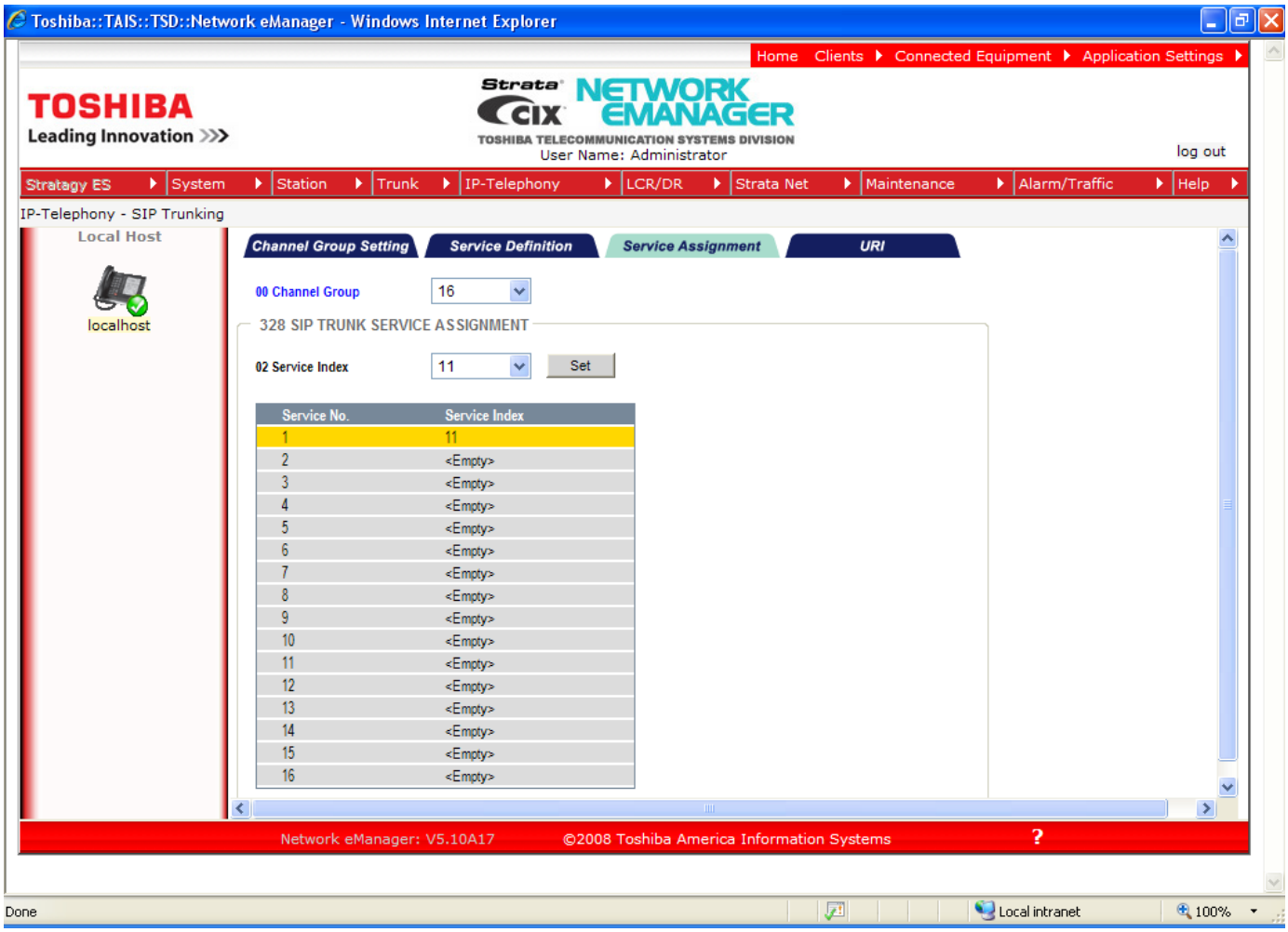

Figure 13 - Service Assignment

URI Programming

The SIP URI is the Telephone Number (TN) from the SIP Trunk provider. The MIPU/GIPU registers each TN using the same authentication credentials (FB03 and FB04.) When the URI TN is dialed from outside, the SIP Trunk provider sends a SIP INVITE just as if it were a call to a SIP station. However, an outbound call using one of the UAs does not "busy out" the UA against an incoming call. That is a key difference of SIP Trunks.

1. FB00 - URI Service Index: The service index that defines the SIP provider.
2. FB01 - Index: Click the Index line in the table to select one of the indices (1-160.)
3. FB02 - SIP URI: This is the TN of the URI.
4. FB03 - SIP URI User Name: **Version N Static**: Leave this field blank. **Version N Dynamic**: "PBX Username" is the BTN listed in the technical welcome letter provided by Broadvox.

5. FB04 - SIP URI password: **Version N Static**: Leave this field blank. **Version N Dynamic**: “PBX Password” can be found in the technical welcome letter provided by Broadvox.
6. FB05 - Channel Group Number - this will be filled in automatically when the **Add** button is clicked.
7. FB06 - Default value is MAIN. If set to SUB, as called out in the URI Programming Table, the URI number cannot be used as the Calling Number in Prog 321.

Important Note: URIs will register as soon as they are Added, Modified, or when the MIPU/GIPU is reset.

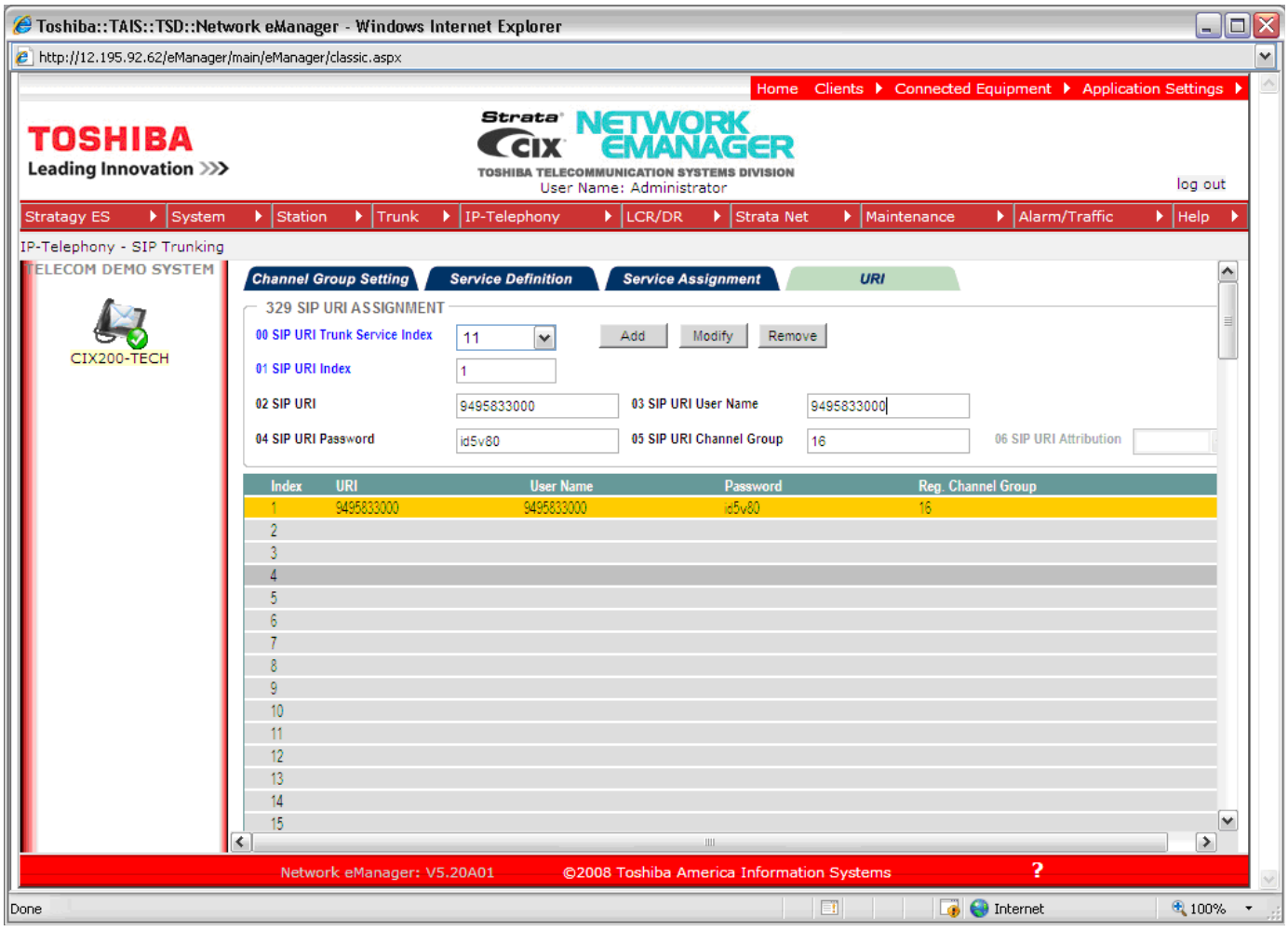

Figure 14 - URI Programming

Configuring the MIPU/GIPU

1. Determine if a static IP address or DHCP (Dynamic IP Address) will be used. This information is available from the IT department at the installation site or from the ISP.

2. Go to IP Telephone > IPU Config. If a static IP address will be used, go to Step 3. If this site is using DHCP, go to DHCP configuration on Page 22.
3. Ensure that FB21 is set to Disable.
4. FB01 - MIPU/GIPU IP address. This value must be a public IP address.
5. FB02 - Subnet Mask.
6. FB03 - Default Gateway Address.
7. There is only one interface on an MIPU/GIPU. Leave the second interface (FB04 - FB06) blank. The DNS server address is critical to resolving the SIP provider's domain name and SIP server. This information was entered in the Service Group Definition. The installation site IT department or the ISP will provide this information.

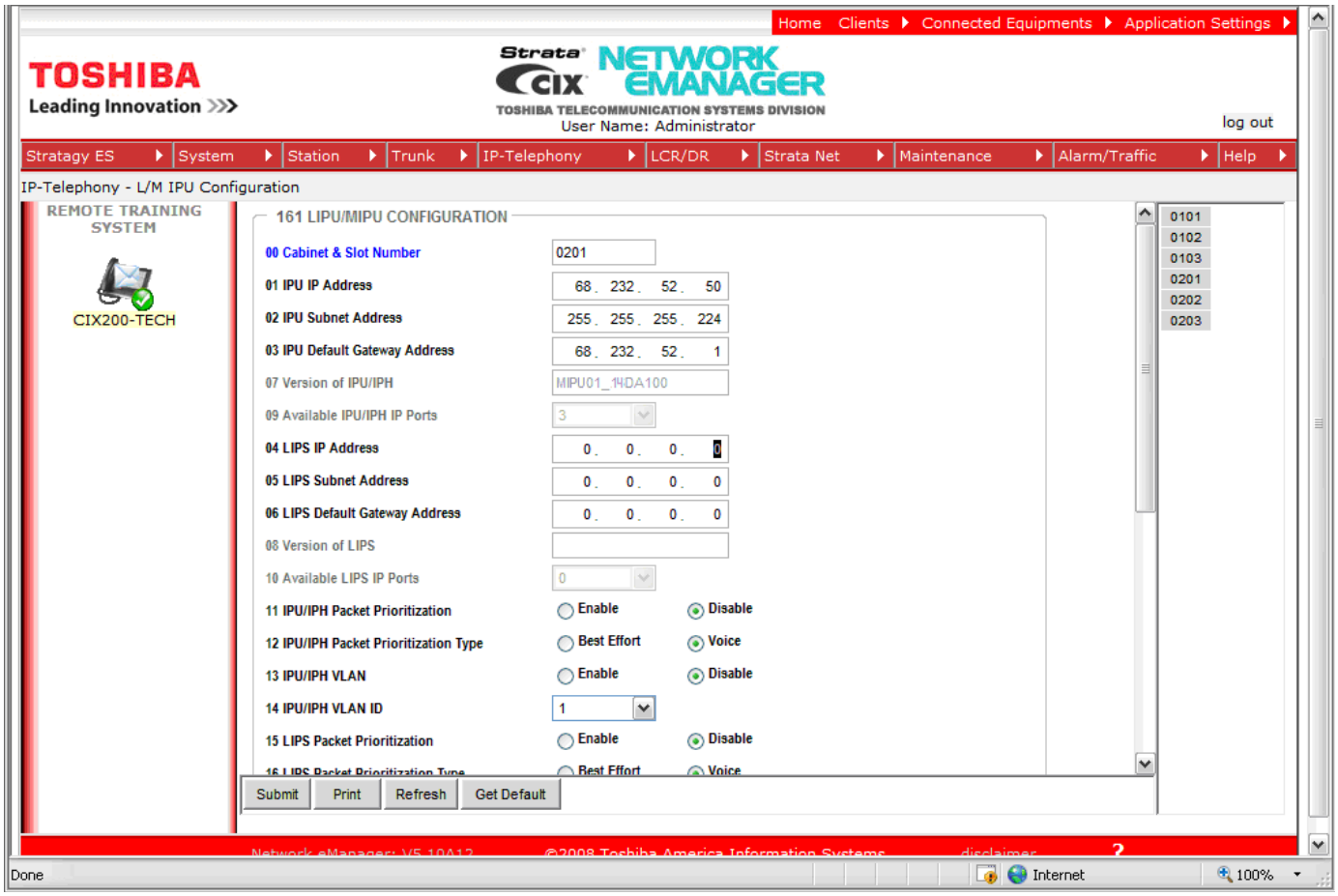

Figure 15 - Configuring the MIPU/GIPU

8. FB22 - Primary DNS IP address
9. FB23 - Secondary DNS IP address
10. Click on the Submit button. Wait for the programming to complete and then go to the next step.
11. Press the **Restart** button on the MIPU/GIPU.

The configuration is complete.

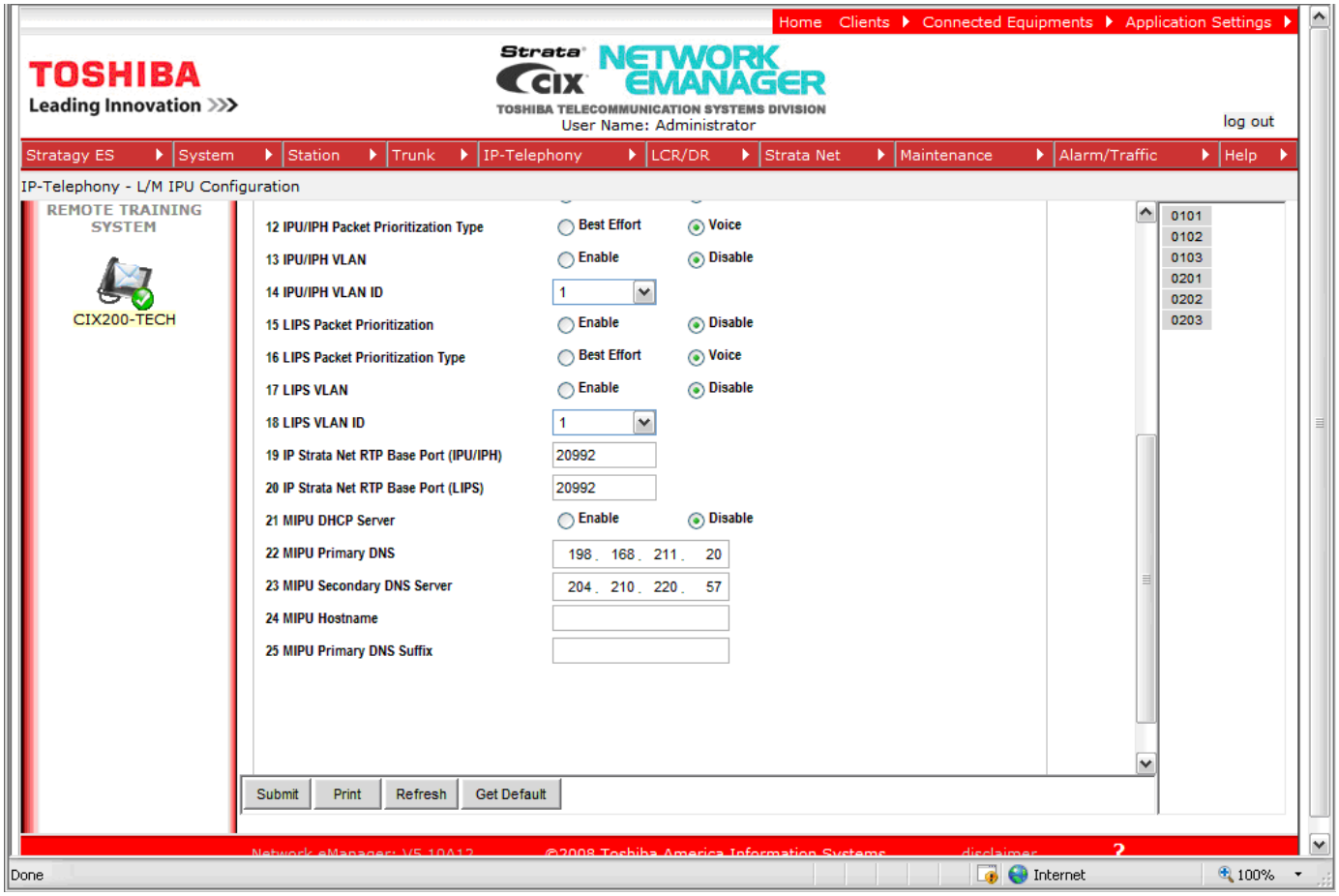

Figure 16 - DHCP Configuration

DHCP Configuration

1. FB21 - MIPU DHCP Server: Click to select Enable.
2. Click on the Submit button. Wait for the programming to complete and then go to the next step.
3. Press the Restart button on the MIPU/GIPU.

The configuration is complete.

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