Call Admission Control

OPERATIONAL DIRECTIONS

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1 GENERAL

1.1 INTRODUCTION

Call Admission Control is used to manage the available bandwidth used for voice calls through low bandwidth links. These types of links are usually used to connect branch offices to headquarters and due to the restricted bandwidth these links normally have, a mechanism is needed to manage them properly in order to maintain a certain QoS in the calls when the media goes through this kind of links. It is also possible to modify the packetization interval of the media (RTP) frames.

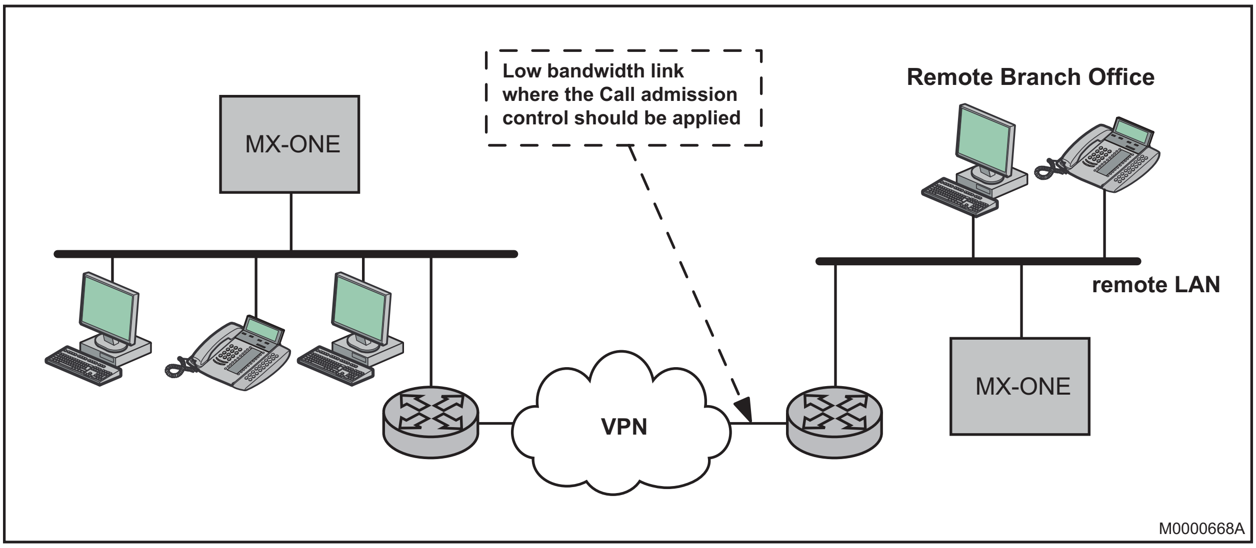

Figure 1: Call admission control scenario

When Call Admission Control is used a maximum bandwidth for all outgoing IP calls (H.323 and SIP) is set per LAN segment. The LAN segments for one area is called an IP domain (which is unrelated to DNS). One IP domain should be set up for each branch office and one for the main office. The lowest common bandwidth will be used for IP communication over these networks.

In Figure the LAN segments A, B, and C belong to the same IP domain, Stockholm. Segments D and E belong to the IP domain Flen and segments F and G to the IP domain Skovde.

For IP traffic within the own IP domain the bandwidth is unlimited.

When defining IP domain take the following into consideration.

CAC evaluates the LAN segment to which the media resource belongs to:

- for the MX-ONE Service Node: it is the IP address of the RTP resource. (i.e. media gateway, configured in *media_gateway_interface*)
- for IP end points (trunk or device): The media IP address in SDP (SIP) or OpenLogicalChannel (H.323).

Note: At outgoing call to an IP end point, when the media address of the target is not known yet (before sending a SIP INVITE or H.323 SETUP), CAC will evaluate the call signaling LAN segment (the IP address with which a device is registered or a trunk is configured (in uristring for SIP or the command RIANI for H.323)).

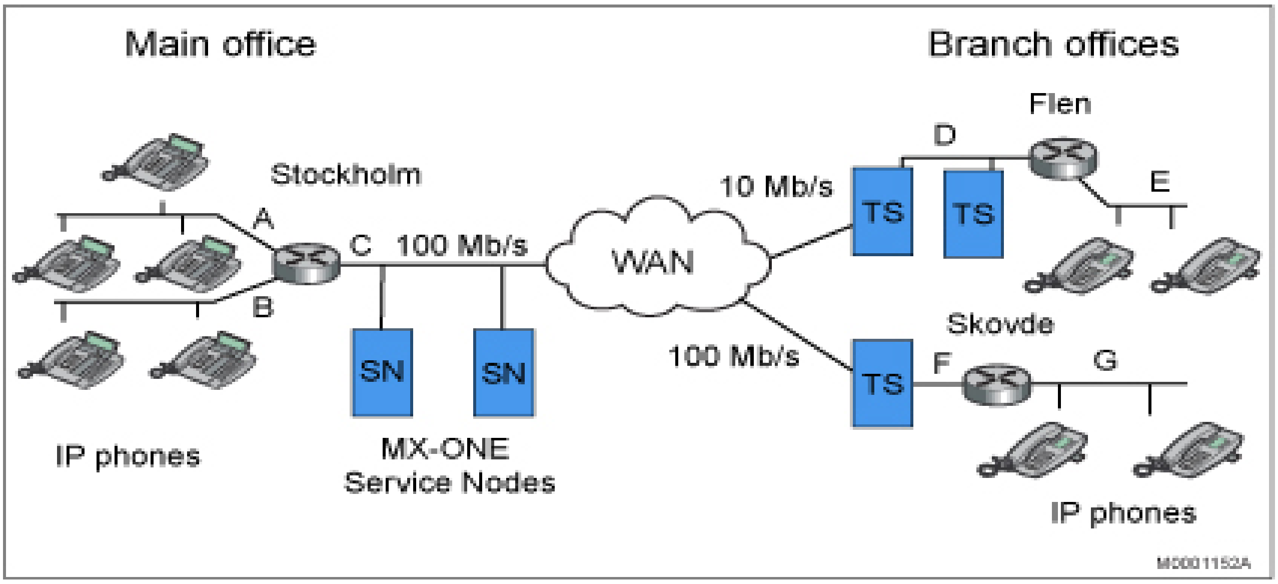

Figure 2: Different bandwidths per IP domain

1.2 GLOSSARY

For a complete list of abbreviations and glossary, see the description for *ACRONYMS, ABBREVIATIONS AND GLOSSARY*.

2 PREREQUISITES

The call admission control facility is closely tied to the configuration of the system:

- The system planning shall have been done for the customer system(s) in question, including the calculations of bandwidth needs, in particular for remote branch nodes (in a different domain than the main system).
- All the IP extensions located in the remote branch office should be registered in the same domain. There should always be IP connectivity between all the IP extensions registered in the same domain.
- If alternative transport methods for media (other than TCP/IP) shall exist, they must have been configured. See OPERATIONAL DIRECTIONS for Transport Connection and related operational directions for Inter-gateway Route.

3 AIDS

I/O terminal.

4 PROCEDURE

4.1 GENERAL

The call admission control only applies for **gateway** calls and **inter-domain non-gateway** calls.

4.1.1 CALL ADMISSION CONTROL IN GATEWAY CALLS

In gateway calls the Call admission control depends on the used codecs and the defined bandwidth.

At IP traffic between domains, where the end point (IP terminal, IP trunk) belongs to one domain and the gateway (RTP resources, configured in *media_gateway_interface*) to another, the endpoint domain take precedence over gateway domain.

The highest priority codec is normally selected for the call, but a lower priority codec can alternatively be selected, in case call admission control congestion rejects a higher priority codec. See the example below where the priority order is shown from left to right.

Table 1 Example of selected codec based on codecs supported by the Media Gateway and terminals

	Available codecs				
Codecs supported by the IP terminal	G.711	G.729a		GSM	G.729b
Codecs in domain		GSM	G.711	G.729b	
IP (SIP or H.323) terminal/Domain codecs		GSM	G.711	G.729b	
Codecs supported by Media Gateway		-	G.711	-	
Remaining codecs			G.711		
Selected codec	**G.711**				

For customers that have upgraded from earlier MX-ONE versions, note that the different Media Gateways support different sets of codecs.

For gateway calls, it is possible to change the priority order of supported codecs by command. If the connected device does not support any of the Media Gateway codecs, the call is rejected. See the example below. For more details on how to calculate inter-Server traffic, see MX-ONE System Planning.

If no codec can be used, the call is either rejected, or if an alternative transport method is configured, the call may proceed via an alternative media connection (e.g. public ISDN). See example below.

Table 2 No common codec can be found

	Available codecs		
Codecs supported by the IP terminal	G.711	G.729b	
Codecs in domain	-	GSM	G.729b
IP (SIP or H.323) terminal/Domain codecs	-	-	G.729b
Codecs supported by Media Gateway	-	-	-
Remaining codecs	-	-	-
Selected codec	-	-	-

Table 3 below shows bandwidth requirements per call direction for some codecs.

Table 3 Bandwidth (for media) for some codec examples, with (default) 20 ms frame length

Codec	Bandwidth, kbit/s (Ethernet)
G.711	54 kbits/s per call direction
Clear channel	54
Unknown codec	54
G.722 (SB-ADPCM)	43 (or 38/32)
G.726 (ADPCM)	22
G.729 A,B, AB (CS-ACELP)	16
G.723.1 (MP-MLQ)	12
G.723.1 Annex C	12
H.264 (Video, unknown bandw.)	5000 kbit/s (or less if specified)

Using ATM instead of Ethernet would increase the required bandwidth by circa 25%. Using encryption also increases the required bandwidth slightly. The trunk configurations and percentage of non-direct-media calls also affect the bandwidth requirements.

If the traffic interest/load is the same between all servers in the system, i.e. there are no remote branch nodes or non-symmetrical distribution of extensions or trunks, then, if it is a large system with many servers, most of the traffic will be between servers.

The internal cases, where calling and called party are registered in the same server, can basically be neglected.

Assuming 2000 IP extensions in two servers (1000 extensions in each), a traffic peak of 0.2 Erlang and standard holding times, the call intensity will be 3.4 calls/second.

This corresponds to circa 200 simultaneous calls, so if G.711 codecs are used, it would be 54*2 kbit/s times 200, which equals a required bandwidth of 21.6 Mbit/s for media. The control signaling adds another 440 kbit/s (or less), to a total of circa 22 Mbit/s.

Note: The customer parameter (if used) has no direct association to the bandwidth. The association is to the domain (IP address range), which should be different per customer.

For more details on how to calculate inter-Server traffic, see *MX-ONE System Planning*.

4.1.2 CALL ADMISSION CONTROL IN A BRANCH NODE SCENARIO, EXAMPLE

For a remote branch node, i.e. a server and gateway located in another domain far away from the main parts of the system, bandwidth limitations are more likely, and the CAC feature is more needed.

Assuming that the remote branch server has 1000 extension users, and 10% of the calls go to/from the main part of the system, with a peak traffic of 0.2 Erlang, and standard holding times, you get 1.7 calls/second, i.e. an average of 0.17 calls/s to/from the main site. The required bandwith shall thus be sufficient for 17 simultaneous calls, and assuming G.711 codecs it equals circa 2 Mbit/s for media, and an additional 22 kbit/s (220/1.7 x 0.17 = 22) for control signalling.

Table 4 below gives an example based on an inter-domain remote branch node case, assuming that all connections use G.711 codec, the peak traffic load is 0.2 Erlang, and 10% of the traffic goes outside the branch node (while 90% is not via the main site).

Table 4 Branch node bandwidth requirement versus number of extensions in the branch node. (With the prerequisites stated above).

Number of extensions in the remote server	Bandwidth, kbit/s
100 (voice extensions)	0.22 Mbit/s
500	1.1 Mbit/s
1000	2.2 Mbit/s

For more details on how to calculate inter-Server traffic, see *MX-ONE System Planning*.

4.2 INITIATION

The following workflow must be followed when initiating the Call admission control feature:

- Configuration of IP-related, system-wide data.
- Initiation of the domain and the bandwidth per codec in the domain.
- Initiation of an IP terminal.
- Configuration of the IP terminal.

4.3 REMOVAL

The following workflow must be followed when removing the Call admission control feature completely:

- Remove the domain.

5 EXECUTION

5.1 CONFIGURE CALL ADMISSION CONTROL

General

The aim of the call admission procedure is to assure that the available bandwidth is not exceeded. The bandwidth per IP domain must be defined.

Note that the lowest common bandwidth for all involved IP domains will be used for IP calls between the IP domains.

Prerequisites

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Execution

1. Key the command *ip_domain -p* to check if the IP domain has already been initiated.
2. If the domain has already been initiated, key the command *ip_domain -c* to change the bandwidth and possibly the codec priority-ordered list for the IP calls. If not, take the next command.
3. Key the command *ip_domain -i* to initiate a network domain name with the included LAN segments and the supported bandwidth for outgoing IP calls.
4. Key the command *ip_domain -p* to verify that the IP domain has been properly defined.

Iterate step 1 to step 4 for all the cooperating IP domains.

5.2 CHANGE IP DOMAIN BANDWIDTH AND CODEC PRIORITY ORDER FOR CALL ADMISSION CONTROL

General

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Prerequisites

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Execution

1. Key the command *ip_domain -p* to list the MX-ONE Service Nodes that are serving the IP domain.
2. Key the command *ip_domain -c* to change the bandwidth for the IP domain and possibly the priority order between the codecs.
3. Key the command *ip_domain -p* to verify the result.

5.3 REMOVE CALL ADMISSION CONTROL

General

The call admission control facility may be removed.

Prerequisites

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Execution

Put the bandwidth to zero. This will restore the default, unlimited, bandwidth.

1. Key the command *ip_domain -p* to list the MX-ONE Service Nodes that are serving the IP domain.
2. Key the command *ip_domain -c* to change the bandwidth to zero for the IP domain.
3. Key the command *ip_domain -p* to verify the result.

6 EXAMPLE

Set up Call Admission Control for the network as outlined in See Figure .. The different LAN segments, marked with letters (A to G) are defined in CIDR format with subnet mask. The default codec priority list is chosen.

ip_domain -i --domain-name Stockholm --ip-net 100.105.10.0/24, 100.105.12.0/24, 100.105.15.0/24 **--bandwidth 100M**

ip_domain -i --domain-name Flen --ip-net 100.130.10.0/28, 100.130.12.0/28**--bandwidth 10M**

ip_domain -i --domain-name Skovde --ip-net 100.132.12.0/25, 100.132.14.0/25 **--bandwidth 100M**

Alternatively set that only codec G.711 u-law, and G729A shall be prioritized for gateway calls to/from and within the Skovde domain. For non-gateway calls that will mean that if the terminals also support other codecs, the G.711 u-law and G.729A will be given higher priority than the other codecs, but the others are still allowed.

ip_domain -i --domain-name Skovde --ip-net 100.132.12.0/25, 100.132.14.0/25 **--bandwidth 100M --codec-priority-list** PCMU, G729A

7 TERMINATION

Inform the department or person responsible for telephony matters if any alteration is made.

If any exchange data have been changed, a dump to backup media must be performed.